00C034I

Project Code ~~HU~~ 00-RGF-JOMP

Joy of Music Program Promotional Sampler Production

An Interactive Qualifying Project Report
Submitted to the Faculty
of
WORCESTER POLYTECHNIC INSTITUTE
In partial fulfillment of the requirements for the
Bachelor of Science Degree
of

Sean M. Blondin

Michael A. Milner

March 2, 2000

Approved by:

Prof. Richard Falco, Advisor

Prof. Douglas G. Weeks, Advisor

Abstract

Working with the Joy of Music Program, the team produced a high quality musical CD sampler portraying their various student and faculty ensembles. Conducting extensive research into standards for location recording led to exhaustive experimentation ascertaining ideal techniques. To ensure quality, professional musician, audio engineers, and graphic artists were consulted throughout the project. The resulting product was used by the Joy of Music for promotional and fund raising purposes, as well as pioneering a relationship between their school and WPI.

Table of Contents

1. ***Introduction***..1

2. ***Literature Review, Interviews, and Observations***...............................6

2.1 **Microphones**..7

2.1.1 **Microphone Selection**..7

2.1.2 **Microphone Placement**..13

2.1.2.1 *Coincident Pair Microphones*..13

2.1.2.2 *Near-Coincident Pair* Microphones..15

2.1.2.3 *Special Near-Coincident Pair Arrangements*..............................16

2.1.2.4 *Spaced-Pair Microphone Arrangements*.....................................17

2.1.2.5 *Special Spaced-Pair Microphone Arrangements*..........................18

2.1.3 **Conclusions**..18

2.2 **Mixing Consoles**..19

2.2.1 **Module Controls**..19

2.2.2 **Tone Control**..21

2.2.3 **Console Connections**..23

2.2.3.1 *Standard Recording Connections*..23

2.2.3.2 *Input Level Adjustments*...25

2.2.4 **Stereo Imaging**..25

2.2.5 **Conclusions**..26

2.3 **Audio Recording Devices**...27

2.3.1 **Analog Recorders**..27

2.3.1.1 *Basics of Analog Audio*...28

2.3.1.2 *Recording* Channels……28

2.3.1.3 *Memory Track Retrieval*……30

2.3.1.4 *Tape and Head Configurations*……30

2.3.1.5 *Maintenance*……31

2.3.1.6 *Degaussing*……32

2.3.2 **Digital Audio Technology……33**

2.3.2.1 *Basics of Digital Audio*……33

2.3.2.2 *Onboard Features*……35

2.3.2.3 *Direct Digital Signal Translation*……36

2.3.3 **MiniDisc Multi-Track Recorders……37**

2.3.3.1 *Onboard Features*……37

2.3.3.2 *Digital Data Compression*……38

2.3.3.3 *Maintenance*……38

2.3.4 **Assessment of Recording Formats……39**

2.3.4.1 *Analog Audio Tape Recorders*……39

2.3.4.2 *Digital Audio Tape*……39

2.3.4.3 *MiniDisc Multi-Track Recorders*……40

2.3.5 **Conclusions……41**

2.4 **Digital Signal Processing……41**

2.4.1 **Digital Filtering……41**

2.4.2 **Relevant Hardware……42**

2.4.3 **Software Capabilities……43**

2.4.4 **Conclusions……45**

2.5 **Interviews with Local Recording Technicians**..............................46

2.5.1 **Joseph Cholorio**...46

2.5.2 **Nicholas Chase and Bobbie Chase**...49

2.5.3 **Brad Pierce**..51

2.6 **Team Observations**..52

2.6.1 **Recording Session on the West Boylston Common**.......................52

2.6.2 **Recording Session at the Assumption College Chapel**...................53

2.6.3 **Recording Session at the Assumption College Chapel**...................53

2.6.4 **Location Recording Seminar**...54

3. ***Methodology***...56

3.1 **Equipment Analysis**..57

3.2 **Rehearsal Preparation**..59

3.2.1 **Bancroft Room Performance Preparations**...................................60

3.2.1.1 *Bass Compensation Experiment*...60

3.2.2 **Sanctuary Performance Preparation**...62

3.2.2.1 *Sanctuary Acoustic Analysis*..63

3.2.3 **Encountered Preparation Problems**...64

3.3 **Production Recording Preparation**..64

3.3.1 **Jazz Ensemble Rehearsal**..65

3.3.2 **Student Orchestra Rehearsal**..67

3.4 **Production Recording**..68

3.4.1 ***Faculty Jazz Recital***...69

3.4.1.1 *Microphone Placement and Monitoring Tactics*..............................70

3.4.2 ***Student Chamber Trio Production Rehearsal*** **74**

3.4.2.1 *Microphone Placement and Monitoring Tactics* 75

3.4.3 ***Student Orchestra Production Rehearsal*** **77**

3.4.3.1 *Microphone Placement and Monitoring Tactics* 78

3.4.4 ***Faculty Chamber Trio Production Rehearsal*** **81**

3.4.4.1 *Microphone Placement and Monitoring Tactics* 82

3.4.5 ***Student Jazz Ensemble Production Rehearsal*** **84**

3.4.5.1 *Microphone Placement and Monitoring Tactics* 85

3.5 **Post-production** **88**

3.5.1 **Sound Editing and Audio Format Transfer** **88**

3.5.1.1 *Example of Sound Forge 4.5* 91

3.5.2 **Graphics Compilation** **92**

3.6 **Conclusions** **94**

4. **Appendices** **96**

4.1 **APPENDIX A: Production Recording Data Log Forms** **97**

4.2 **APPENDIX B: Discussions Concerning Sampler Graphics** **102**

4.2.1 **Graphics Discussion with WPI Communications Group** **102**

4.2.1.1 *Initial Plan of JOMP's Sampler Compact Disc Encasement Elements* 102

4.2.2 **Initial Meeting with A Graphic Artist** **103**

4.2.3 **Initial Meeting with Wendy and Richard Ardizzone** **104**

4.2.4 **Final Meeting with David Blondin, and Mr. and Mrs. Ardizzone** **106**

4.3 **APPENDIX C: Sampler Template Mockups** **107**

4.4 **APPENDIX D: Examples of Graphic Editing** **108**

4.5 **APPENDIX E:** **Final Sampler Cover Graphic Templates**...............**109**

4.6 **APPENDIX F:** **Sampler Text Templates**...................................**111**

5. **Bibliography**...**115**

Chapter 1

Introduction

Location recording is an art form that captures a brief moment in time and preserves it eternally. Capturing a live performance through recording requires technological knowledge and artistic acumen. Having an understanding of recording equipment, musical nuance, the location's acoustics, musicians' needs, and audience perspective is the key to successfully recording a musical production. The final audio product should emulate qualities of past great location recordings, while retaining the uniqueness of the specific performance.

This Interactive Qualifying Project (IQP) team's mission was to capture live performances of the Joy of Music Program's (JOMP) students and faculty through location recording. The musical selections from the JOMP ensembles' repertoire were made available to a wide audience through the mass production of a sampler audio compact disc produced and designed by the IQP team. The goal was to produce a CD of near professional quality.

Located at the First Unitarian Church of Worcester, the Joy of Music Program (JOMP, pronounced "jump"), founded by Wendy Ardizzone in 1986, was initially created to introduce preschoolers to a world of music and dance. Remaining a nonprofit organization since November of 1989, and currently under the direction of Wendy and Richard Ardizzone, JOMP now offers children, adults, and families the opportunity to develop talents and skills in music and dance. The school is a member of the National Guild of Community Schools of the Arts enrolling over 550 students annually, a tremendous expansion over its original 12 members. Its broad curriculum includes private instruction, music and movement classes, and beginner to advanced jazz and classical performance ensembles. JOMP's thirty faculty members provide a professional learning environment for aspiring musicians.

As many recitals are held throughout the year, JOMP's classes are designed to give students experience in live performance. Its students, faculty and guest artists presented 48 performances in its 1997-98 academic year, reaching a combined audience of over 6,000 people. JOMP's Chamber Orchestra, jazz ensembles, chamber music groups and Afro-Caribbean drumming ensembles have performed at many community events including: First Night Worcester, The Worcester Convention Centre Opening, The Worcester Mayoral Inauguration Celebration, and the Annual Meeting of the Greater Worcester Community Foundation. Its Chamber Orchestra also performed live on WICN's classical radio music program "Serenade".

As JOMP's goal is to prepare students for live performance, the CD sampler produced by the IQP team portrays the JOMP ensembles in a performance environment. However, this is not comparable to a studio production, where all conditions can be controlled to achieve flawless recording. The musical selections included on the IQP team's sampler were recorded on location, where there existed many variables and limitations inherent in the physical performance space where the recordings took place.

Five of JOMP's ensembles were recorded in two rooms located at the First Unitarian Church of Worcester. The Bancroft Room hosted recordings of a chamber student trio and two jazz groups, one student jazz ensemble and a faculty jazz sextet. Worcester's First Unitarian Church sanctuary hosted recordings of a full student orchestra and a faculty Baroque trio.

To achieve a near-professional product while dealing with location limitations, exhaustive preliminary analysis consisted of literature research that explored the proper use of WPI's equipment, which presented limitations when compared with "ideal" professional equipment. Topics of research included the study of hall acoustics, equipment selection and operation, natural and synthetic effect compensation, and stereo imaging.

In addition to literary investigation, invaluable interviews of professional recording technicians were conducted. Interview questions explored the capabilities of the IQP team's available equipment and how it could best be used to properly project the live sound from JOMP's performance ensembles onto a standard audio CD. Professionals selected for interview included Joseph Cholorio, Nicholas Chase, Bobbie Chase, Brad Pierce, Charles Paquette, Professor Frederick Bianchi, and Jason Boudreau.

Supported by accurate, up-to-date literature and the advice of these recording technicians, equipment experimentation and rehearsal recordings helped to reveal the most promising of the IQP team's experimental recording methods. Hypothesized recording methods were tailored specifically to capture each ensemble in their respective performance hall. From the most favorable results, a presentation was produced and reviewed by JOMP staff and independent professional musicians who were familiar with critiquing recorded music, to determine which particular method most properly reproduced the original performance sound.

The final technical procedures of post-production required the assistance of an audio technician for computer-aided transfer from the original master recordings to the sampler compact disc prototype. Other post-recording work included the completion of the sampler's cover art and liner notes by working closely with photographer and graphic artist, David Blondin. Sound editing and graphics production were adjusted to the preferences of Richard Ardizzone, JOMP's director of this production.

As the final goal of this IQP, the project team presented the Joy of Music Program with its first sampler compact disc of faculty and student performances including graphics and liner notes outlining the school's purpose and goals. As a result of the IQP team's work, JOMP prepared its sampler for mass production. Copies were distributed to radio stations, parents of performing students, sponsors of JOMP, and sold to raise funds for the advancement of music education through JOMP scholarships. The final sampler exceeded JOMP's expectations of a near-professional quality consumer product.

Chapter 2

Literature Review, Interviews, and Observations

Literature Review

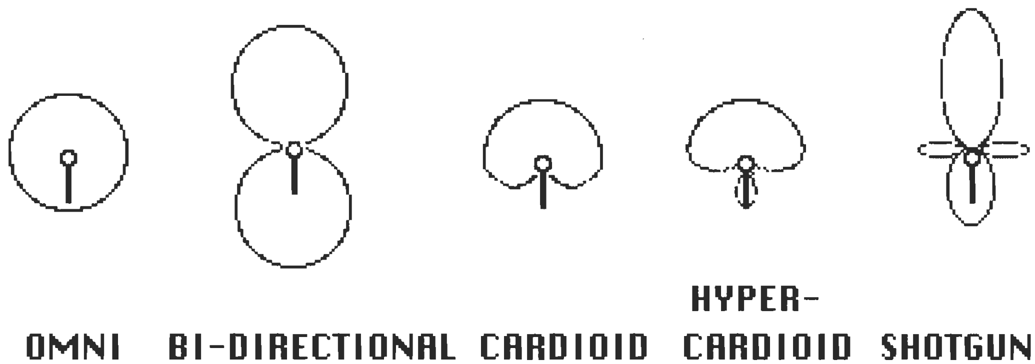
2.1 <u>Microphones</u>

Initial sound capturing is vital to location recording since, unlike studio recording where post-production product enhancement is emphasized, it is difficult to improve the characteristics of a poorly recorded live performance. Special attention must be given to the microphones, which is the first component into which the sound is captured. Choosing the correct microphones depends on whether they will be used for studio or location recording, their defining characteristics, and the desired qualities of the final product. Microphone choice cannot rest solely upon industry reviews since defining characteristics of the recording subject must be considered in the decision process. Microphone characteristics such as frequency response, sound sensitivity region, and optimal room location must be investigated.

2.1.1 **Microphone Selection**

There are two main types of microphones, dynamic and condenser (Bartlett, 4). Both dynamic and condenser microphones have a thin membrane, called the diaphragm, which translates received physical vibrations into electronic signals using a transducer (Traylor, 29). Dynamic microphones require no active electronics to amplify the input signal from the sound source. However, condenser microphones require either batteries or a voltage known as *phantom power* sent through the transmission line to power their internal electronics (AKG Manual). Condenser microphones have a broader frequency response, but they are more fragile and more expensive. Dynamic microphones are general-purpose microphones, which have a more focused range and are more limited in frequency response than condenser microphones (Olson, 141).

Since recording subjects and environments vary greatly, there are many characteristics of microphones that allow more adaptability. The shape of the three dimensional field in which a microphone receives signals depends on the type and model of the microphone(s) used. The standard sensitivity regions, or pickup patterns, are *omnidirectional*, *unidirectional, bi-directional,* and *cardioid.* Pickup patterns are usually depicted as polar diagrams, which are circular graphs of a microphone's range of sensitivity to sounds produced from various directions. Each of these polar responses allows the specific microphone to be used in a specific way in specific situations.



The microphone pattern commonly used for recording ambient sound is the *omnidirectional.* An omnidirectional microphone pattern should follow the spherical sensitivity range as described by Olson (Olson, 68). As its name implies, this pattern is sensitive to sound signals from all directions, including behind the microphone. This pattern is a very popular choice when natural reverb is desired, as with orchestral and chamber recordings. These microphones lack proximity, as they capture so much ambiance that a clear, focused source is lost. This is undesirable with jazz recordings, but highly desired with choral groups, where blending and reverberation are primary concerns.

Unidirectional microphones, on the other hand, are designed to focus on a direct sound source to capture less ambient signals. The definition of "unidirectional" can apply to other patterns as well, so the term also refers to a highly focused microphone called a *shotgun*. The pattern is a narrow cone emanating from the front of the microphone. These are primarily used for special recording purposes such as interviewing and capturing point source sounds (Huber, 222).

Bi-directional microphones are also special purpose microphones. Bi-directional microphones have a figure eight pattern created by two spheres on either side of the sensitivity regions (Huber, 67). These two spherical regions can be oriented to the sides of the microphone body or in the front and rear of the microphone, depending on the manufacturer. This pattern was originally designed for vocal duets, but is now used in certain stereo placement techniques.

In general, the most popular microphones have a *cardioid* pattern. Cardioid microphones have a heart-shaped pickup pattern. As shown in the previous diagram, they reject sound from the rear, which reduces, but does not eliminate, ambient sound. Cardioid microphone tone is governed by a phenomenon called the Proximity Effect. This describes the increase in bass as the microphone moves closer to the sound source. Similarly, the further a cardioid microphone is from a sound source, the more high frequencies will be accentuated, creating a treble-dominated tone (Huber, 174).

Unpowered, dynamic microphones are usually labeled as omnidirectional, meaning they are sensitive to sound all around the microphone, but the region immediately in front of the microphone is most sensitive and this region is where the microphone captures most of the sound (Huber, 57).

Dynamic microphones can be either magnetic or piezo-electric. The latter is reserved for inexpensive microphones used for low quality sound capturing (Traylor, 25). High quality dynamic microphones use magnetic elements to convert changing magnetic fields produced by the diaphragm's vibrations into electric signals (Olson, 77).

For dynamic microphones, JeepJazz, an organization that specializes in live jazz recording, suggests the Shure SM57 as a general-purpose microphone. This microphone is well known to be the optimum singer's microphone, as its limited range is well suited for capturing sound close to the source. According to sound engineer Bruce Bartlett, similar choice microphones are the Crown CM-311A, Shure SM81, AKG-D112, Sennheiser MD-421, and the Beyer M88. Bartlett based his study on scientific qualities such as frequency response (the frequency range in which the microphone is sensitive) and quality of sound.

Duncan R. Fry, author of Live Sound Mixing, reports that the Shure SM58 is the industry standard for close micing. His personal choices as comparative microphones are the Electro-Voice N/D757 or N/D457, Beyer M88, Sennheiser 441, and Shure SM57. His list is based more on personal opinion than scientific data, as he is a live sound engineer working with touring bands. The frequency response bandwidth of dynamic microphones is limited, so their primary use is for close placement to the source (Refer to interview with J.Cholorio, 46). Utilizing this quality, individual sources can be controlled exactly with minimal ambient noise.

The drawback of using this close microphone recording technique is the restrictive sonic range that a singular microphone can capture. In order to capture the sound of a complex instrument such as a drum kit, multiple microphones must be placed to capture each individual drum. In fact, most professionals use 8-10 microphones on a drum kit (Fry, 241). In addition to recording a drum kit, each other instrument must be captured using individual microphones. Therefore, for a small jazz band of two horns, bass, guitar, piano and drums, most professionals, including Fry, would usually use over 15 microphones (Fry, 234). The microphone mixer must then be capable of merging these many channels into two stereo channels, assuming the recording medium is stereo. From the array of resources, most professionals agree that this use of dynamic microphones is typically reserved for studio recording. One important element, as WPI music Professor Fredrick Bianchi states, is that live recording needs to be reproduced exactly as an audience member would hear it. Constant monitoring and minute adjustments of microphone levels, which are commonplace with multiple microphone use, should not happen during a live performance recording, as this is not what an audience experiences. Using a large number of channels also requires intense skill as incorrect balancing can produce an artificial and undesirable sound (Refer to interview with J.Cholorio, 46).

From the majority of the resources, most professional sound engineers, including Joseph Cholorio, Prof. Fredrick Bianchi, and Bruce Bartlett agree that live recording should be done with condenser microphones. Although the physics of a condenser microphone is very similar to that of a dynamic microphone, the output signal is produced much differently (Traylor, 31). A condenser microphone does not produce the voltage that is transmitted to the preamplifier usually contained in the mixer. In condenser microphones, a static charge is created between the diaphragm and a plate behind the diaphragm. The vibration of the diaphragm alters the gap between the two, fluctuating the capacitance, which in turn creates a fluctuating electrical current. Because the diaphragm membrane must be much thinner than those in dynamic microphones, they are much more sensitive and capture a more realistic sound (Refer to interview with J.Cholorio, 46).

Using two microphones allows the entire ensemble to be blended together with room acoustics, allowing the listener to experience the nuances of being part of the audience. A popular type of condenser microphone is the stereo microphone. A stereo microphone is a microphone case containing two microphone elements, giving the dual microphone effect. The two elements are often angled anywhere from 0° to 270° from each other (Bartlett, 69). These angles are determined by the theory of phase cancellation. When recording in stereo, there usually is a certain amount of signal that is transmitted to both channels. If the signal received in one microphone is delayed by 180° with respect to each other, meaning the wave is essentially the same frequency, but inverted amplitude, the opposite amplitudes subtract from each other (Bartlett, 45). This phenomenon occurs with central sources. Therefore, these microphones must be strategically placed. To compensate for placement limitations, these angles can be adjusted accordingly (Olson, 67).

Bartlett's scientific research demonstrates why his choices for the best stereo microphones are the AKG C-426B, Audio-Technica AT822, Schoeps MSTC 64 "ORTF", Shure VP88, and the Sony ECM999. These microphones, being rather expensive, come with many options ranging from adjustable incident angles, a variety of polar patterns, and selectable sensitivity levels. Additionally, a stereo microphone uses only one microphone stand, which is more aesthetically appropriate in a live performance environment than multiple stands. Since stereo microphones are out of budget for the average sound technician, two identical microphones properly positioned can achieve the same stereo effect as a single stereo microphone (Bartlett, 34).

2.1.2 **Microphone Placement**

To emulate a costly stereo microphone, two condenser microphones can be used to simulate the desired stereo effect (Bartlett, 45). Many different microphone positioning techniques are used for capturing the source without encountering the phase cancellation problem previously mentioned. Many of these methods are based on the Cartesian plane where, to capture the most sound, the microphones are placed one over the other with the source located at 90° and the microphones are skewed equiangularly from the 90° axis (Bartlett, 81).

2.1.2.1 Coincident Pair Microphones

Another family of microphone techniques is called the "coincident-pair", which is a pair of matched microphones with their grills, the screened surface of the top of the microphone coinciding at the same point, as shown on the previous page. The bodies of the microphones are then angled from one another to create different effects. Because the sound is captured at a focal point, the point where the grills meet, the stereo quality is diminished. However, there is also no phase cancellation because the minuscule spacing between the capsules. An asset of this system is that if desired, the recording can be monaural compatible even if stereo panning is done (Bartlett, 72). The angle of the bodies to each other determine the stereo separation, with small angles producing a monaural effect since they will be receiving similar signals, while angles larger than 60° will result in a stereo distinction between the two channels.

The following configurations are specifically designed to be implemented using microphones with cardioid patterns. The first configuration consists of two cardioid microphones placed one grill over the other with tails 180° apart. This configuration places the recorded reverberation to the extreme left and right channels and dulls direct sound. This method also produces a cavernous sound that is used when heavy reverb is desired. If heavy reverb is desired with more focus on the original source, a smaller angle is necessary. A smaller angle between 120°-135° allows some of the original source to reinforce the indirect sound captured by the wide spacing method (Gerson, 45).

A study done by Huggonet and Jouhaneau, shows that using the Blumlein or Stereosonic Technique, two coincident bi-directional microphones angled 90° apart, produces the most uniform possible spread of reverberation and the sharpest focus of any angle (Bartlett, 112). Adjusting the microphones to approximately 90° allows a more realistic blend of ambiance and direct signal.

Another coincident-pair technique is the Mid-Side technique, known as MS (Bartlett, 121). This technique uses a middle central microphone capsule aiming toward the source combined with a bi-directional microphone placed so the two fields of the bi-directional stereo microphone intersects the cardioid pattern of the middle microphone. This method is highly customizable and can cover a broad spectrum of situations and sound sources. Balancing the focus between direct signal and ambient signal respectively allows balancing between these two without changing the microphone location during recording. This method is very monaural compatible. A study by Ceoen and Griesinger demonstrates that the MS method actually creates an artificial sound due to its lack of room acoustics (Bartlett, 124). Similar techniques include using hypercardioid microphones with tails angled 110° apart (Bartlett, 119).

2.1.2.2 Near-Coincident Pair Microphones

A second family of techniques is the near-coincident pair method. Stereo microphone recording in this manner consists of two cardioid microphones angled much like the coincident pair method, but with the grills being at a greater distant, thus forming an 'X' with the microphone bodies (Huber, 54). The benefit of this method is greater stereo differentiation, leading to an improved, realistic sound. This method, according to Bartlett, is not monaural compatible because if the two channels were to be mixed at equal levels, phase cancellation would occur because corresponding frequency would be received on either microphone at slightly different times which can decrease the amplitude of certain high frequencies (Bartlett, 45). However, as the two capsules are moved closer to each other, phase cancellation is less likely to occur, as in the coincident pair technique.

2.1.2.3 Special Near-Coincident Pair Arrangements

The near-coincident method is widely used in stereo recording because of these characteristics, but variations on the near-coincident microphones' placement create additional techniques. One method is known as the NOS system developed by the Dutch Broadcasting Foundation (Bartlett, 122). In this situation, a matching pair of cardioid microphones is placed in a V pattern at 45° to each other with an 12" distance between the two grills, creating a large gap between the grills, which face outward from each other. This distance extension, in contrast to the ordinary near-coincident pair arrangement, creates a deeper focus of the left and right channel boundaries. The distance can be hyper-extended if necessary by separating the two microphones and extending the gap to 16". This further increases the delineation between each channel (Bartlett, 78).

Another interesting method is the OSS (Optimal Stereo Signal), or the Jeckin Disk method. This is the placement of two omnidirectional microphones spaced 16.5cm (6.5") apart and separated by a perpendicular foam covered disc with a diameter of 28 cm (9") (Bartlett, 123). This method attempts to mimic the natural stereo effect observed by human ears. Humans perceive a sound in one ear before the other. This delay allows a listener to perceive depth and location of a particular signal. Because the nature of this method optimizes listening to the recording through headphones, playback through a pair of speakers may not necessarily have the same perception quality.

Another popular method is the ORTF, developed by and named after the French Broadcasting Organization (Bartlett, 80). Two cardioid microphones are angled 110° apart with grills spaced 17 cm (6.7") apart. Any angles greater then 110° lose central focus and the sound becomes weak, and any angles much less then 110° lack peripheral intensity (Bartlett, 83).

A technique called the Stereo 180 System is yet another near-coincident pair method. Developed by Lynn T. Olson, it uses two hypercardioid pattern microphones angled 135° and spaced 4.6 cm (1.8") apart grill to grill (Bartlett, 116). This method receives a limited response pattern that creates the illusion of a narrow space that simulates a studio situation.

2.1.2.4 Spaced-Pair Microphone Arrangements

Another common microphone placement method is the spaced pair technique (Huber, 125). This method is a relatively simple method that is used for general recording purposes. Audio technician Joseph Cholorio suggests using this method mainly for orchestral recording (Refer to interview with J.Cholorio, 46). Spacing the microphones 3' apart gives an accurate central focus, but weak off-center sources. This method is often used with other microphone pairs for additional emphasis on particular sounds to the stereo mix (Bartlett, 84). A 10' spacing is a prime method for recording an orchestra because of the wide region this space covers (Refer to interview with J.Cholorio, 47). The distance from the source is a major concern when using the 10' spacing, as many signals can be lost from the middle of the ensemble (Traylor, 43). The spaced pair method is a method that varies immensely with respect to the size and location of an ensemble in a given environment. Experimentation with a variety of microphone positions is the only way to locate the optimal implementation of this method.

2.1.2.5 Special Spaced-Pair Microphone Arrangements

A special method reported only by Bartlett is a method developed by Telarc Records which uses three omnidirectional microphones placed in a linear pattern 5' from each other to compensate for the potential loss of central signal (Bartlett, 119). A similar method by Decca Record Company also uses the 5' distancing, but with the central microphone placed closer to the source than the two peripheral microphones (Bartlett, 121).

2.1.3 **Conclusions**

By interpreting the results from the literary review and interviews, it was concluded that condenser microphones have the optimal characteristics for live recording. Not only did professionals recommend condenser microphones along with their respective techniques in placement, but scientific evidence also reinforced condenser microphones as the prime choice for the type of location recordings this IQP team performed.

The spaced, coincident, and near-coincident stereo pairs emerged as the most promising of the microphone placement techniques. Recording in a room of great overall ambiance, as with any large hall suitable for orchestral performances, requires a microphone arrangement that portrays the ambiance of the actual performance. The spaced pair was selected as the First Unitarian Church sanctuary's assigned microphone placement technique for its reverberant and spacious sounding stereo imaging. Near-coincident and coincident pair microphone arrangements, although less defined in stereo substance, are usually used at close distances when recording smaller ensembles arranged in small performance environments. These two methods were chosen as the assigned techniques used for performance recordings in the Bancroft Room of the First Unitarian Church. These techniques were adjusted according to which ensemble was being recorded.

2.2 Mixing Consoles

The mixing console, sometimes referred to as "board", "desk", or "mixer", is an audio control panel that consists of several identical modules, running parallel to one another. Individual inputs from musical instruments and microphones are plugged into these modules. Each of the modules can either be muted or actively assigned to one of the mixer's two output channels for a stereophonic product.

2.2.1 Module Controls

The typical mixing module consists of five main controls used for adjusting the separate channels for the mixing process. The sound adjustments most commonly found on mixers are *gain trim*, *auxiliary output*, *equalization output*, left and right channel *panning*, and *fade*. Separate from module controls exists the output level display that allows for manual balancing of the two final output stereo channels. According to Bruce Bartlett, the definitions for each of the module controls are given as follows:

The *input trim* or *gain* regulates the input signal level from each microphone to the desired output level. The *auxiliary* controls are normally used to govern the amount of effects on each tape track, but can also be used for other balancing purposes.

Equalization (EQ) is the term used to denote an intentional change in relative amplitude response at different frequencies. This is commonly referred to as tone control. The dial for low-end frequencies, which range approximately from 20 to 150 Hz, controls the bass tones. The mid-bass dial, a frequency feature defined between low-end and midrange, controls frequencies at levels between 150 to 500 Hz. Controlled with its dial, midrange frequencies are found within the boundaries of 500 Hz to 5000 Hz (5kHz). The dial for the top-end frequencies controls the treble tones, roughly ranging from 5000 to 20000 Hz (5 kHz to 20kHz).

The *pan* controls allow the separate input channels to vary its assigned microphone signal output to either or both of the left and right channels as desired. Signal output may also be represented more on one channel than the other.

The *fader* acts as a sliding or dial positioned volume control, depending on the mixer brand and type. This control has the special ability to affect or bypass the line output level, which will be discussed later.

2.2.2 **Tone Control**

For the final mixed output, each input signal can be adjusted as desired via the cooperation of properly arranged microphones and balanced EQ dials. It is important to become familiar with the mixer's EQ by manipulating specific frequency dials, one at a time, while observing how they enhance, justify, and complement each other. Depending on how one frequency tone is adjusted with respect to another, many synthesized acoustical qualities can result: *liveliness*, *intimacy*, *fullness*, *clarity*, *warmth*, *brilliance*, *texture*, *blend*, and *ensemble.*

Liveliness is the term used to describe a room's reverberation time. Reverberation is the most important component of hall acoustics. It is caused by the reflection of sound waves off hard, reflective surfaces, commonly known as echo. Reverberation is defined as the duration of time between the first heard direct sound and the last heard reflected sound or, defined as, the amount of sound decay. The more reflective surfaces a room contains, the more reverberant characteristic it has. A room is said to sound more "live", when it has a longer reverberation time. A room lacking reverb is said to be "dead" (*Analysis of Spaulding*, 14).

Intimacy is the term that describes the illusion that a listener feels he or she is listening to the performance in a small hall and is surrounded by the ensemble's sound (*Analysis of Spaulding*, 14). This phenomenon is achieved when the first reflected sound reaches the listener in less than 20 milliseconds after the direct performance sound. A hall is said to deliver a *full* sound when the reflected sound level is of similar intensity in comparison to that of the direct performance sound. This could also be explained as the intensity of the hall's reverb properties.

In opposition to intimacy and fullness, a room with less intense reverb is said to have *clarity.* A room of fine performance clarity is desired for musical pieces having a fast tempo so the performance ensemble can deliver distinct notes that are unlikely to blur with the reverb of the hall. In most halls, some frequency ranges will have more or less reverberation time than others. When reverb favors the low-end (bass) frequencies the performance projects a *warm* sound. It is said that a room becomes *muddy* with excessive warmth. In the opposite case, when reverb favors the high-end (treble) frequencies, the performance sound produces a *brilliance* effect. In the case where brilliance becomes too great, a high-pitched ringing may be evident during the playback of the final recording.

Beranek best describes *texture* as "*the subjective impression created in the mind of the listener by the pattern in which the sequence of sound reflections arrives at the listener's ears*" is dependent on the ratio of sound reflections per time interval. A room with texture of significant quality should produce 5 reflections per 60 milliseconds following the direct sound (Beranek, 70). It is important for the sake of quality texture that each successive reflection is of lesser intensity than the previous for proper decay (*Analysis of Spaulding*, 15).

It is also important for each listener within a hall to hear a strikingly similar, if not identical, combination of direct and reflected sound, regardless of location in the room. This refers to a hall's degree of *blend.* Significant blend refers to level of similarity between the listener's received sequence of source sounds and the sequence of direct source sounds. In a hall with exceptional sound blending, all audience members will receive the instrumental or vocal sounds in the sequence in which they are directly performed. In a room with less significant blending qualities, accurate diffusing surfaces may need to be erected around the performance to achieve desired blend (*Analysis of Spaulding*, 16).

Another important characteristic of any performance hall is *ensemble*. Ensemble is the ability of a performer to hear other performers in direct sequence with relationship to the location of each musician. To achieve ensemble, the performance area's reverberation time must not be longer than the fastest notes played. This suggests that the blending quality in the performance area should be of the highest standard and, hopefully, is also identical to the blend reaching the audience (*Analysis of Spaulding*, 16).

2.2.3 **Console Connections**

The purpose of connecting a multi-track recording device with a mixing console is to record the signal of each microphone onto an assigned tape track. Separation of the tracks is not only useful when attempting to create synthetic performance imaging during playback, but it also creates added flexibility when performing post editing using software programs or other sound editing consoles.

2.2.3.1 Standard Recording Connections

A mixer is an essential component of any recording process, location or studio, because it contains pre-amplification for each microphone input. Preamplifiers built into each input module allow a multi-track recording device to be fed a line-level signal, which is a signal that is intense enough for the recorder's input meters to detect (Bartlett, 8).

The line-level signal is normally sent out from two connectors found on the rear of the mixing console: *direct out* and *insert send.* This is where the tape-track inputs can be connected. Because the *direct out* jack is post-fader (comes after the fader), the signal is affected by the mixer's fader settings. In this situation, any fader movement will transfer to the recorded tape, which is undesirable. It proves much more practical to connect tape tracks to *insert sends*, which are normally pre-fader. In this situation, the main recorder's input levels will ignore any fader movements. This method is useful when additionally running the mixer's *direct out* into a Public Address (PA) system, as it allows the PA operator to adjust the mixer's faders and the PA's volume without affecting the recording device's input levels.

On the rear of the mixer, the *phantom power* switch supplies the microphones' active electronics with a 48-volt source. This voltage source is required for the activation of the microphones' electronic capabilities to convert acoustic waves into electronic modulations. The electronic signals are sent to the mixer's input via XLR cables and later manipulated using the mixing controls.

The XLR cable is the standard connection between all recording devices. This balanced cable connector contains three electronic conductors. They are designated as shield, audio in-phase (hot), and audio opposite polarity (cold). While the physics of their placement is based on sophisticated electromagnetic calculations, the three conductors are so wound and well shielded that they carry audio signal without interference from external sources of magnetic and static anomalies such as speakers, power-lines, and fluorescent lights (www.recording.hostway.com). XLR cables provide an interference-free communication medium between microphones and mixer inputs, mixer insert sends and main recorder channel inputs, and main recorder channel outputs to secondary recording device channel inputs.

2.2.3.2 Input Level Adjustments

As a performance is being recorded, it is important to properly adjust both the mixer's and the recorder's meter levels. This is accomplished by either adjusting the trim or gain controls on the mixer or by regulating the input levels on the recording device. It is important to make certain that all respective channels are adjusted to approximately the same input level on the recording device. An average recording level of -10dB (decibels) is desired (Bartlett, 9). Since the input level clips, forcing the recorded sound to distort, at a reading of 0 dB, the -10dB input setting creates a buffer zone that allows for any sudden level escalations.

2.2.4 **Stereo Imaging**

Imaging is the result of receiving sound source signals at two different points, namely the left and right ear. A signal will reach each ear at slightly different times with variations in amplitude. Although these time lapses may last only microseconds and the amplitudes may vary by millidecibels, these subtleties give the listener an audio image of where the sound sources are located. Because these variations depend on the interaction of the subject with the geometry of the location, special care must be taken with microphone placement and channel panning in order to capture the stereo image into two separate channels.

Recording at least two separately positioned microphones mixed through two properly arranged channels, panned left and right, allows for instrument imaging when listening to the playback of the recording through a stereo system. Imaging becomes even more realistic when a stereo recording is heard through a pair of headphones.

Since scientific means of determining the optimal stereo microphone arrangement is limited, as each performance environment is unique, experimentation on location is the most practical means of analyzing the stereo effects of a particular technique.

2.2.5 **Conclusions**

The Bancroft Room and the First Unitarian Church sanctuary have unique sound qualities, some of which were not desired on the final promotional disc. Equalization output levels were not adjusted during the performance recordings, as many literary resources and technical professionals including Jason Boudreau, Joseph Cholorio and Professor Bianchi have discouraged tonal manipulation during location recording. The equalization controls were set flat (at the neutral level) in advance of each recording session. Regarding this, manual definition or suppression of specific frequencies was necessary during post-production. The team could have used a mixer during the process of transporting the master recordings onto the compact disc audio format in order to set proper equalization levels of all frequencies. However, further sound editing was needed beyond the mixer's equalization capabilities.

Knowing that mixers have great limitations when used in post-recording sound editing and that the final production had to be as near to professional standards as possible, the IQP team completed post work with a more versatile software program (See **Post-production** section, 88) in which many more frequencies and effects were adjusted. But for the purposes of pre-amplification capabilities and its simple two channel output controls, a standard two-track input/output mixing console worked well in conjunction with the team's two-track stereo recording device.

2.3 Audio Recording Devices

Magnetic tape, digital tape and the more recently introduced high-density digital computer discs, more commonly known as compact discs (CD's), provide the current standards in audio recording mediums. The two tape formats, analog (magnetic) tape and digital tape, share a negligible number of characteristics. Analog tape systems operate by recording continuous signals where as digital tape systems function by recording a minimum number of signal samples. Technicians are also able to record using the newly introduced MiniDisc (MD) multi-track format, a form of digital technology that utilizes a portable disc system. All three alternative mediums have advantages and disadvantages in respect to each other.

2.3.1 Analog Recorders

"We can define "analog" as something that is similar or comparable to another in certain respects. Calling a device an analog audio tape recorder (ATR), then, refers to its ability to transform an electrical input signal into

corresponding magnetic energy that is stored in the form of magnetic remnants on tape." (Huber, 101)

2.3.1.1 Basics of Analog Audio

Professional analog ATR's utilize standard cassette or reel-to-reel recording systems. The compact cassette based models are modern in comparison and are limited to recording a maximum of 4 tracks. The much larger reel-to-reel units are generally configured in 2, 4, 8, 16, and 24-track formats. Each of the tracks contained within the ATR is suited to perform a specific task in recording production and postproduction. The number of tracks available on the ATR should be no less than the number of potential microphone line-inputs. (Huber, 112)

It is imperative that either a 16-track or 24-track ATR be used during a multi-instrument recording session for studio style recording. In the case where a drummer must have the snare, toms, bass drum, and cymbals separately recorded, or where one or more musicians use more than one instrument throughout a single performance, there may be a possibility that more than 16-tracks will be needed. There is also the choice of using more than one ATR if more than 24-tracks are needed, but this complicates post-work editing and format transfer.

2.3.1.2 Recording Channels

An analog recorder's magnetic tape head is specialized to perform three tasks: record, reproduce, and erase. Its electronics are arranged to aid the operator in manually instructing these functions. In modern ATRs, regardless of track configuration, the input circuitry is duplicated identically for each channel. A single mainframe inside the ATR's console houses the input/output (I/O) module. Multi-track machines employ an I/O module in which all of the adjustment controls and single channel electronics are incorporated on a single printed circuit board. The I/O modules allow for adjustments to input levels, output levels, synchronization levels, and equalization. An advantage to this design is greater channel interchangeability and service ease. (Huber, 113)

An ATR's output signal may be switched between three modes of operation: *input mode*, *reproduce mode*, and *sync mode*. The *input mode* sends a signal immediately from the line-input to the channel output. This enables the capability to meter and monitor all signals, directly from the input, independent of any ATR transport mode.

The *reproduce mode* forces the output channel to read directly from the reproduce head. In this mode, all meters and monitors will reflect the playback signal. This mode is useful in two ways in that it allows for the playback of previously recorded tape, and it enables monitoring of the record mode signal while not actually recording. The second of the two features can be examined while an ensemble test performance is underway so as to adjust for undesired effects. This allows a recording tech not only to prepare the recording levels using the level meters but also to hear and then reduce any unwanted anomalies from background noises, "dirty" connection cables from corrosion or wear, and faulty hardware.

The synchronization, or *sync,* mode allows the recording of one or more inputs onto one or more tape tracks while listening in synchronization to the playback of previously recorded material. This is commonly referred to as the *overdubbing* process. A professional ATR supports the capability of independent switching between the record and playback functions for each channel. A safety switch should also be included with this feature to avoid accidental erasure of a recorded track (Huber, 115-116).

2.3.1.3 Memory Track Retrieval

Another convenient feature supported by ATR systems is the *auto-locator*. This feature allows specific recording times to be stored into memory. When a technician retrieves a specific cue point from memory, the auto location feature will shuttle the tape to the desired position. This allows any portion of the recording to be assigned a specific time in which the technician can then proceed with the necessary examination or manipulation of the desired portion of tape. This feature is not a necessity, as some ATRs do not support the auto-location advantages, but it is useful in saving the recording technician a lot of valuable time (Huber, 117-118).

2.3.1.4 Tape and Head Configurations

According to Huber's Modern Recording Techniques (118), analog ATRs are available in a broad spectrum of track-width and tape-width configurations. The most familiar configurations are 2- track, ¼"; 4-track, ½"; 8-track, 1"; and 16-track and 24-track, 2".

Optimal tape-to-head performance is dependent upon two parameters: track and head-gap width and tape speed. A greater track width promotes greater magnetic sensitivity, resulting in an enhanced output signal and improved signal-to-noise ratios for each recorded track. The unrecorded strip of tape found between recorded tracks, called the *guardband*, prevents channel signal overlap.

Tape speed is directly related to both the recorded signal's level and wavelength. The number of magnetic elements that move past the tape head gap during a set time interval becomes greater with increased tape speed. Faster tape speeds generate greater average magnetization recognized by the playback head, resulting in stronger signal definition requiring less amplification and, therefore, less tape noise.

A faster tape speed also allows for a greater recorded bandwidth, resulting in increased high-end (treble) frequencies. This is due to the equality found between the signal frequency cycle and head gap during high speeds. As the frequency of the playback signal increases, more and more of the complete frequency cycle will remain inside the boundaries of the head gap, until the gap width, at any one point in time, is equal to the signal wavelength. When this is accomplished, the average output level will become zero. Reducing the output to a level of zero results in *scanning loss*, which is the determinant of the upper frequency limit. With less limitation, more of the high-end frequencies are detected. When low-noise, high-output tape is incorporated, further noise reduction processes are unnecessary (Huber, 119).

2.3.1.5 Maintenance

It is a necessity to protect the magnetic recording heads and mechanical components of an ATR transport deck from dirt and oxide shed. Oxide shed occurs when friction between the tape and the ATR's mechanical components causes small particles of magnetic oxide to accumulate onto the tape surface contacts. This accumulation should be most carefully looked for at the surface of the magnetic recording heads. If excessive oxide shed occurs here, the separation between the tape and the head can cause minor or significant signal loss, both of which are undesirable. For instance, a signal recorded at 15 inches per second, with an oxide buildup of 1 mil (0.001") at the playback head will yield a playback loss of 55 dB at 15 kHz (Huber, 120). Huber also suggests using denatured alcohol or an appropriate cleaning solution at least daily and always before routine alignments (Huber, 120).

2.3.1.6 Degaussing

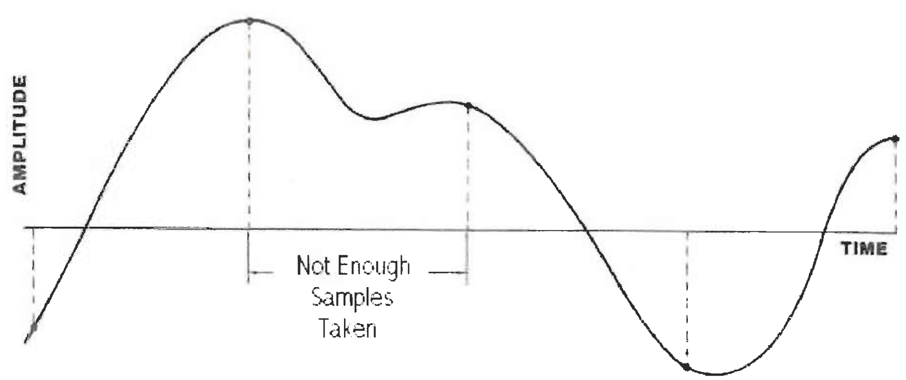
Another strongly suggested method of ATR maintenance is the degaussing practice. Since the magnetic tape heads are composed of a "magnetically soft" metal alloy, the alloy, although functioning as a great conductor of magnetic flux, does not readily retain magnetism. Nevertheless, these heads do retain small, but not negligible, amounts of residual magnetism that can partially erase the high-frequency signals on a master tape. For this reason the degaussing of the magnetic tape heads within 10 hours of operation is strongly recommended. A magnetic head degausser acts very much like an eraser in that it saturates the magnetic head with a very high-level alternating signal, which randomizes the residual magnetic flux. Once the head has been degaussed, it is important to remove the degausser from the tape heads at a speed of less than 2" per second, so as to avoid inducing a magnetic flux in the head (Huber, 121).

2.3.2 **Digital Audio Technology**

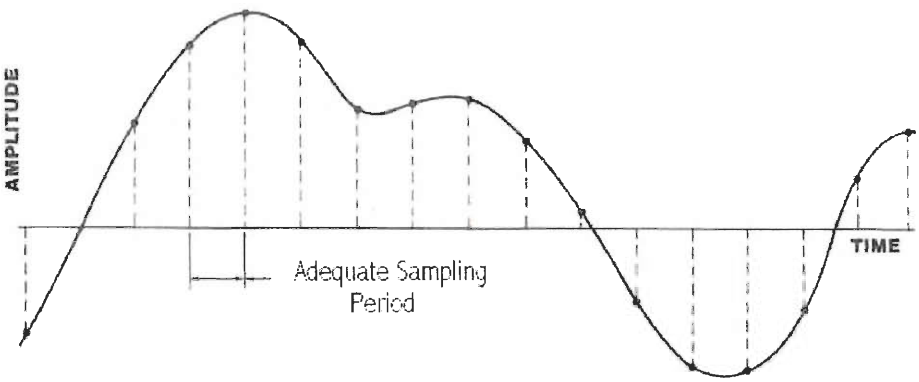
> *"Over past two decades, digital audio has evolved from an infant technology, available to only a few, to its present position as a primary driving force in audio production. Digital audio has affected all aspects of the audio industry, home and studio." (Huber, 129)*

2.3.2.1 Basics of Digital Audio

A modern technique for recording is using the relatively new digital format. Whereas analog technology supports the recording, storage, and reproduction of the natural, uninterrupted changes in signal level, digital recording does not function in a continuous manner. Taking periodic *samples* of the same natural, continuously changing waveform replicated by analog recorders, a digital audio tape (DAT) recorder can organize sampled signal levels into a representative set of binary numbers, storing them for later exact or edited reproduction.



If too few samples are taken from the original continuous signals, a distorted signal is produced, leaving only scraps of the original signal to be heard.



When enough samples are taken, digital reproduction can be an extremely accurate system while requiring less storage than an analog system. If too many digital signals are analyzed the result is a much larger digital database with no greater detail than the previous representation, containing a smaller but equally accurate database.

A digital audio system's *sampling rate* refers to the amount of digital samples taken from a source signal per second. The *sampling time* is the reciprocal of the sampling rate, representing the time between each sample period. Bandwidth is directly dependent upon the sampling rate. A higher sampling rate yields a wider bandwidth.

The DAT sampling process samples an incoming analog signal at a fixed sampling rate in which each recorded sample is examined momentarily to represent its assigned voltage level. These sampled levels, then, undergo an important digital mathematic conversion process resulting in a sequence of binary number sets, each of which represent the value of sampled signals at each time interval in accordance with the DAT's constant sampling rate.

The *Nyquist Theorem* states that the minimum sampling rate achieving maximum accuracy must be at least twice the value of the highest recorded frequency. Most digital audio recorders have selective sampling frequencies of 32-kHz, 44.1-kHz and 48-kHz. When analyzed with Nyquist's theorem, most recordings require the DAT minimum frequency limit selections should be 16-kHz, 22.05-kHz and 24-kHz respectively (Huber 145).

2.3.2.2 Onboard Features

DAT is designed to equal or exceed many standard digital specifications. Its capabilities boast highly sensitive signal reception, wide dynamic range, and low distortion. Employing an enclosed compact cassette, smaller than that of a standard compact audiocassette, DAT players are equipped with both analog and digital input/outputs. Able to record and reproduce at all previously mentioned sampling rates, a DAT must be set at either the 32-kHz or 48-kHz sampling frequencies when recording from an analog source. The intermediate 44.1-kHz sampling frequency is reserved for the recording of prerecorded DAT tapes and compact discs. The 44.1-kHz and 48-kHz sampling modes offer a 2-hour running time, where the 32-kHz sampling rate offers three record/reproduce modes. The first option provides 2 hours of recording time with 16-bit linear quantization, the second option provides 4 hours of 12-bit nonlinear quantization, and the third option offers a 4-hour recording of four-channel, nonlinear 12-bit audio (Huber, 145-146).

Like the ATR, each DAT system has its own level meters, tape location time counter, and auto-location techniques. DAT recorders also support a unique feature that allows for track-time assignment and recognition in which the play button may be pushed, during a recording session, at a desired time to assign that time interval a new track number on the tape. This system works not unlike standard consumer compact disc players, in that when wanting to reach a specific time interval on the DAT, the tracks saved onto the DAT tape are read by the DAT for the ease of consumer track-referencing. Playback a DAT recording at a specific track or time interval is made simple with this feature.

2.3.2.3 Direct Digital Signal Translation

Over the years of its development, digital audio has inspired many software companies to create DAT compatible sound editing programs. The first programming step for digital audio integration has been to allow the placement of audio into computational digital storage for the further manipulation of digital data known as sound "files". This process was easily developed since DAT recordings and computer files are both comprised of digital writing and digital retrieving operations. The transfer of DAT information to digital computer files is said to be a flawlessly reproduced translation. These sample-based audio systems are programmed to move, copy, edit, and replace sonic data.

A more descriptive substitute for the term sample-based audio is the term *random-access audio system*. All computer files, including audio files, are capable of being accessed in random order. This allows a technician almost instant access of desired "blocks", or "sound file" sections, without having to sift through an entire performance file. Some current random-access systems include samplers, signal-to-disc transfer, multi-channel digital audio workstation, and the two-channel digital audio workstation.

2.3.3 **MiniDisc Multi-track Recorders**

> *"Highly efficient and multifaceted, the minidisc multi-track recorder is the low-flying but extremely practical Swiss Army knife of digital home recording: Basically put, it makes good-sounding digital recordings at an analog price." (October 1999 edition of "Home Recording": Gelfand, 48-49)*

Designed to combine the advantages of modern professional recording platforms, the innovative MiniDisc (MD) multi-track platform boasts the self-contained portability of an analog cassette unit with near-CD-quality audio presentation.

2.3.3.1 Onboard Features

All professional minidisc systems support an onboard recording devices and multi-channel mixing console including inputs for instruments and microphones, gain trim and equalization controls, variable pitch control, send and return effects loops, in addition to headphone jacks and other controls. While these features are also supported by many ATR platforms, the MD units also include disc searching, editing (copying, moving, and song segment reordering) features, jog/shuttle dials for accessing these features, and Musical Instrument Digital Interface (MIDI) interactivity (Gelfand, 49). The integration of MIDI technology enables a user to use multimedia computers and electronic musical instruments to add effects to recorded material during or after production (http://www.midi.org).

2.3.3.2 Digital Data Compression

One of the MD's unique features is its capability of digital data compression, which allows for the maximization of disc storage space. A listener may find this technological innovation to be a useful feature because of its portable attributes, but it has no advantage over the recording time boundaries offered by ATRs and DATs. When set to its highest quality recording mode, an MD is set to the least degree of compression. As a result, compression and degree of sound quality are inversely proportional to one another and create a trade off between recording time and signal definition. Because the MD's recording disc is so small in size, it can store no greater amount of its highest quality digital data than a standard CD (approximately 74 minutes), and far less in contrast to the recording tapes used in analog and digital tape systems.

The unwanted result of MD digital compression is a recording that lacks in bandwidth, sacrificing the lowest and highest audible frequencies that are essential for a hiss-free digital representation. When the compression settings are positioned to higher levels, the bandwidth becomes smaller causing a lack of high and low-level frequency sensitivity. Gelfand comments that although recording with MiniDisc prevents a technician from producing a full-bandwidth recording, the resulting quality still remains "quite good" (Gelfand, 49).

2.3.3.3 Maintenance

The MD unit also permits thousands of hours of proper operational use without the need of maintenance. "*You'll be up and running on an MD unit in less time than it takes to degauss your old cassette multi-track's dust-covered heads.*" (Gelfand, 49)

2.3.4 **Assessment of Recording Formats**

ATR, DAT, and MD recording platforms have been analyzed. The review and comparison of each format and related characteristics, together with the parameters of the IQP team's requirements, resulted in the appropriate format selection.

2.3.4.1 <u>Analog Audio Tape Recorders</u> (ATR)

- *Advantages*
  - Available in a compact cassette platform, useful for easy transport and recording fewer tracks, or reel-to-reel formats used for the recordings requiring more than 4 tracks and up to 24 tracks.
  - Some units are available with built in mixing consoles.
  - Reel-to-reel units provide hours of recording per tape-roll.

- *Disadvantages*
  - Analog tape has noticeable limitations with high-end and low-end frequency ranges.
  - Reel-to-Reel models are cumbersome to transport.
  - Special models must be acquired for memory and auto-locate capabilities.
  - Frequent cleaning and degaussing maintenance is strongly suggested, due to *oxide-shed*. This will require more of the IQP team's time spent on careful attention of material unrelated to the actual production requirements.
  - Less compact than either the DAT or MD units.

2.3.4.2 <u>Digital Audio Tape</u> (DAT)

- *Advantages*
  - More compact than reel-to-reel ATR systems.
  - Represents the highest degree of digital quality.

- Digital enhanced bandwidth. Noticeably versatile frequency response in comparison with ATR and MD limitations.
- Standard features include instant playback, track assignment, and track location, most of which are not available on many ATR units.
- Provides at least 2 and up to 4 hours of recording per tape.
- Flawless transfer onto other digital mediums including other digital tapes and random access computer files, both of which are potential candidates for post editing.
- No machine maintenance is suggested.

- *Disadvantages*
  - Slightly larger than analog cassette units and MD units.
  - No built in mixing capabilities on most units; pre-amplification required for line-level adjustment.

2.3.4.3 <u>MiniDisc (MD) Multi-Track Recorders</u>

- *Advantages*
  - Extremely compact, as a mixing console is usually built in.
  - Incorporates digital data compression technology, which conserves storage space.
  - Presents near-CD quality.
  - Flawless digital data replication during transfer.
  - Standard features include instant playback, track assignment, and track location, most of which are not available on many ATR units.
  - Requires no maintenance under normal use.

- *Disadvantages*
  - High compression settings cause a greater loss of bandwidth.

- Recording time is no greater than the length of a standard compact disc, approximately 74 minutes.
- An MD's recording quality is inferior to that of a DAT system.
- Referring to a recent Gelfand reference (see page 39), the highest MD quality is still "quite good". Well, "quite good" may not be included in the realm of professional sound quality.

2.3.5 **Conclusions**

Regarding the previous "compare and contrast" arrangement of the three formats in question, it is clear that the Digital Audio Tape recording format has the most advantages and least disadvantages. DAT systems are specialized for the highest quality sound reproduction in 2-track recordings. The DAT format remained as the IQP team's constant recording medium for all location recordings throughout the project. In agreement with the team's decision, professionals such as Professor Bianchi, Joseph Cholorio, and Jason Boudreau encouraged the team to use a DAT system in conjunction with a two-track mixer and a stereo pair of condenser microphones. (Refer to **Interviews,** 46)

2.4 **Digital Signal Processing**

> *"Benefiting from the ability to equalize without analog filters prevents distortion commonly caused by component imperfections." (Traylor, 34)*

2.4.1 **Digital Filtering**

Digital filtering is accomplished through shifting of the binary components as recorded by the DAT (Lathi, 234). Since binary is the language of computers, they have become the primary resource for digital editing and digital signal processing.

Digital editing is a relatively new method. Years ago, digital signals had to be converted to analog using analog filters, known as band-pass filters, whose function limits the amplitude of each frequency range (Olson, 43). The analog signal needed then to be converted back to digital format. This method does not have the advantages of modern digital signal processing (DSP) since the limitations of analog processing remain. Computer programs have been designed to accept the digital input from a DAT player and store it as a file on the computer's hard drive. Once this file is created, the digital data can be manipulated more thoroughly then any analog means.

Some of the functions which can be performed in digital format are filtering, equalization, compression or expansion of dynamic range, time compression or expansion, delay, reverberation, pitch change, generation of arbitrary signal or noise, music and voice synthesis, noise reduction, signal restoration, automatic pattern and voice recognition, time-reverse, noise gate, automatic gain control, mixing of signals, and Fast Fourier Transforms (FFT) (Lathi, 356).

2.4.2 **Relevant Hardware**

These programs are primarily used on a Macintosh computer, as the original IBM based computers were not powerful enough to handle the enormous size of the digital file. Today, the standard Macintosh's Small Computer System Interface, or SCSI, hard drive interface still is more powerful then the IBM's ISA interface with respect to speed and accuracy. Most developers design programs specifically for the Macintosh, and then modify them for the IBM architecture (Computer, 44). Another benefit of the Macintosh is the implementation of large amounts of Random Access Memory, or RAM. This RAM is used to store information currently being manipulated. When a file is being read, a segment of the file is always being loaded into memory before the user hears it. This is referred to as a buffer. Since the entire piece of music may be loaded into this buffer, the Macintosh's quality and quantity of memory modules are highly desired. IBM-style computers are closing the quality gap by incorporating SCSI interfaces with newer computers.

When an IBM-style computer, commonly referred to as the personal computer, or PC, receives a digital signal from a DAT, it is converted to a file format called *WAV*, while Macintosh uses a format called *AIFF*. These formats are similar, but incompatible because of the Microsoft/Macintosh issues. These files can be 8, 16, 32, or 64 bits –simply put, the higher the number of bits, the more accurate.

2.4.3 **Software Capabilities**

Products for computer based editing are ProTools, Cubase Audio, Sound Forge, Cool Edit, ReBirth, Acid, Cubasis Audio 1.6, Cakewalk Pro 8 and Jamma Pro. All of these programs allow basic editing tools previously mentioned in 2.4.1, but many allow additional modules to add custom features. Some features a sound technician may require are declicker/denoisers that can eliminate unwanted sound from a recording. Punching is another module that allows a certain sound to be removed from a recording. This method along with other effects such as flanging and chorus are exotic for this specific live recording purpose.

The most standard and widely used program is ProTools, created by Digidesign (Mix, 17). This program has the facilities for very accurate digital manipulation of the balance and tone. This program is specifically designed to operate with a ProTools 24-bit sound card, a custom component to interface the DAT input with the computer. Digidesign has released several versions of the ProTools program and the sound card to cater to specific recording and editing needs. Generally, ProTools is used for post recording editing, but can be modified for direct recording. The benefits to ProTools, as well as many other programs, allow previewing of a modification to the file before it is permanently changed. This is not easily accomplished with analog because audio monitoring must be done from the final product as it is recording during the transfer from DAT. ProTools is also the industry's standard for digital post production because of its reliability, accuracy, and customization packages.

ProTools is primarily used on Macintosh computers and PC's with an Intel Pentium microprocessor using Windows NT (http//www.digidesign.com). This severely limits its use for non-professional digital editors. Other programs can run on any platform under any operating system. Another limitation of digital editing is the cost of these programs. Since these are very specific and very precise programs, the price ranges from several hundred to thousands of dollars for the basic package. The expansion modules can run in the same price range depending on the quality and amount of options each module contains, as well as for which program the module is designed.

Fortunately, most other computer programs mimic the user interface and functionality of ProTools. By studying the characteristics of ProTools through literature and experience in using the program, a user can adapt to other high quality editing programs like Sound Forge, Sound Designer and Cakewalk. These three specific software packages are available in demonstration format with limited features.

2.4.4 **Conclusions**

Much of the technique related to using computer-based digital signal processing is obtained from experience and advisement from professionals, as few books have been written on these programs. Since the nature of computer programs is dynamic, with programmers releasing bug fixes, updates, and patches daily, current literature of a specific program is limited. Much current information is found on internet newsgroups (large databases where individuals upload and download information) and in current professional periodicals. Much of the literature to be found explaining certain modules and product use come from these periodicals, along with the most recent manuals of available programs. In fact, hundreds of popular programs are only documented within their help files, which are included and accessed within the program itself.

Interviews

2.5 Interviews with Local Recording Technicians

2.5.1 *Synopsis of Interview with Joseph Cholorio of Mechanics Hall Productions. Conducted on October 12, 1999 via telephone.*

Joseph Cholorio is the sound technician for Mechanics Hall Productions located in downtown Worcester. He has over 30 years of experience in recording live performances.

What methods are common for testing balance between direct and reflected sound?

Before starting, there should be an objective sound, usually found from previous recordings of the style of music to be recorded. Once a certain base sound is established, adjustments can be made to try to simulate the professional recording. One method of recording seeks to maximum direct sound, usually a desired approach with a jazz band. The assumption is that the room will provide a small amount of acoustic reflection that is desirable to provide ambiance that is characteristic of live recordings. Another method used for classical recording is practice recording an ensemble on location without an audience (i.e. rehearsals). The objective is to find a good location for balancing direct with reflected sound focusing on the room's characteristics and not compensating for an audience. Then experimenting with relocating the microphones to reduce reflected sound. Having knowledge of these locations will help develop optimal ratios of direct-reflected sound when a live audience is present. Still, the decision must ideally be made beforehand - as few adjustments can be made during the actual performance recording.

Do you recommend using a compressor on location? What else do you recommend for unwanted side effects?

Although compressing signals from solo microphones seems appropriate, compression is used primarily in studio work. During live performance, dynamics are desirable, but not typical 'booms', 'clicks' and other undesirable sounds which occur if the microphones are improperly used by the soloist. If any, minimal compression should be utilized if the user of the microphone understands proper use. Usually the stereo microphone pair is already adjusted to capture the sound of a soloist who is playing louder then the accompanists, so a solo microphone is not always useful.

What are the most common techniques used with recordings in church sanctuaries, as the acoustics become more complicated with high ceilings and vast dimensions than in a common room?

The size of the room only affects the ambient aspects of recording. The size and location of the ensemble is more important to microphone placement. Most orchestras are recorded with a stereo pair configuration known as the space pair. The individual microphones are placed at a distance over 3 feet apart. Frequently, the spaced pair is placed trisecting the length of the orchestra. This method centralizes the sound by dual reinforcement from the overlapping range of the two microphones localized at the center of the ensemble. The centralization diminishes toward the extremes as just one microphone receives these sounds. This mimics the way an audience member would perceive the orchestra. The division between the extremes via the right/left channels is known as stereo spread and is important to give a realistic balance to the recording. Very often, 10-11 foot microphone stands are placed in the front row of the sanctuary 8-10 feet apart. Adjustments for exact locations and positions will have to be made to accommodate the position and size of the orchestra.

What microphones are best when only stereo microphone placement is used?

Condenser microphones must be used in order to capture the dimensions of a large ensemble. Dynamic microphones would be unable to capture the spread to the *necessary –60dB level required for a quality recording. Using a cardioid pattern microphone reduces direct sound from the audience because of the general shape*

What microphone placement reduces the amount of bass resonance when the room causes poor acoustics in the lower frequencies?

Eliminating the reverberation is the best measure to eliminate this undesirable quality. The coincident or near-coincident pair located close to the source prevents a great deal of bass reverberation from the naturally bright sound characteristic of condenser microphones.

How would you go about editing in post-production?

Most often, Pro Tools is used. This software package allows a great deal of digital editing impossible with the analog format. The amount of editing depends on how much the recording warrants. Excessive natural reverb, excessive background noise, unwanted sounds from the ensemble are factors that need to be addressed with editing. Because many of these problems can be filtered out to a great extent with Pro Tools, too little of the presence of these nuances make the recording sound artificial by removing the natural experience of being at the performance. If a performance hall has a certain characteristic that is acceptable, this nuance should be included.

Frequently, overproduction occurs by those who have little training. Adding too many effects like reverb, frequency filtering, phasing and other effects can add to the quality of the sound in very small amounts. However, too many effects may make the recording interesting, but unnatural. The goal with digital editing is to enhance the live experience for the listener. Therefore, just the most basic effects should be used in digital editing, such as fade in/out and light filtering.

2.5.2 ***Synopsis of Interview with Nicholas Chase and Bobbie Chase, local location recording specialists.***
Conducted on October 14, 1999 via telephone.

Mr. and Mrs. Chase have been recording live concerts for WICN's radio broadcast for over ten years. They are highly experienced with recording in Worcester performance halls including the First Unitarian Church sanctuary and Bancroft Room, where this IQP's recordings were completed.

What method would you use for recording an orchestra?

A tall spaced pair of condensers have always provided enough signal to produce a quality recording. No tone editing should be involved during the recording. The difference between Nicholas and Bobbie Chase's recording styles is that Nicholas adjusts the gains to prevent inadequate or excessive dynamics. His reasoning for this is to optimize for WICN's broadcast. He chooses to bypass the radio station's automatic compressors by manually adjusting gain levels during the initial recording. Bobbie Chase records primarily for production, so she never adjusts the levels unless clipping is imminent.

In addition to the two condenser microphones, Nicholas Chase uses another condenser or dynamic microphone placed where a soloist or announcer would be performing. This microphone is usually trimmed to zero signal, but is turned up when clarity is necessary in case the entire ensemble is overpoweringly loud. This microphone is mixed in with the right and left channel for central imagery. The goal is to record the performance with all its nuances, so little, if any post-recording work is done.

Since most of our research regarding technique has already been corroborated through this interview, what suggestions do you have regarding recording as far as preparation and setup for the recording?

The most important area is intense preparation for the unexpected. Since the methods for studio recordings don't apply with location recordings, new setup methods and other considerations need to be followed. The unpredictability of the performers, audience noise and movement, performance location, legal protocol specific of the music to be performed music and power issues must be investigated before the actual recording takes place and dealt with properly.

There are many other aspects to location recording than just the act of recording. One consideration is the fact that unlike a studio, there are people all around, walking about, coughing, babies crying, talking and causing other problems that effect the quality of the recording. The first method to dealing with these problems is to locate the microphones close to the performance and as far as possible from the audience. Usually, this is accomplished by raising the microphones as high as possible and aiming down. Since this can eliminate only so much of the ambient noise, the audience should be informed that the performance is being recorded and they should be courteous and avoid making excessive noise.

There are also other sources of noise that usually go unnoticed initially, but which will result in a tarnished recording. Heaters turning on, clocks ticking, other church related functions such as meeting groups and other music ensembles practicing and outside noises such as traffic usually are not noticed when setting up for a recording because they are natural in life, but are undesired on musical recordings.

Another problem encountered during recording in high traffic locations is the possibility of audience members tripping or unplugging power or audio cables. Microphone stands can be knocked over if the cable is jerked because the ends are locked into the hardware and the electrical plug can become dislocated from the outlet. A method to prevent these accidents is taping cables to the floor and taping plugs to their outlets. The tape will prevent excessive movement in case the cables are stepped on.

2.5.3 ***Synopsis of Interview with Brad Pierce of Audio Palace Recording Studio. Conducted on October 8, 1999 via telephone.***

Brad Pierce is the owner and chief audio technician of Audio Palace Recording Studio located in downtown Worcester. Although his specialty is studio recording, his experience with location recording and digital sound editing provide an additional view on the processes to be used.

What methods would you use for location recording a duet or solo performance in either the Bancroft Room or Sanctuary of The First Unitarian Church?

Close micing the individual instruments, mixed with an ambient microphone would work best. Using dynamic microphones for the close micing and condensers for the ambient sound works well. The close microphones can be condensers, but their gain must be reduced. The tone may be 'brighter', but can be compensated for by using minute tone adjustment. Another consideration is the number of each type of microphone necessary for each recording. This is why having several condenser microphones is important.

How would you master a location recording done in the Bancroft Room/Sanctuary?

Many location recordings contain sounds that are undesirable like coughing, conversations, and wrong notes especially towards the very end of a piece. Since these sounds may have too much presence in the final product, careful filtering with Pro Tools is possible, but many times, the sound cannot be eliminated without ruining the original recording of the performance. Usually to eliminate a short sound, the sound is mixed again with an inverted sample of the sound superimposed over the original recording. Sometimes this cancels enough of the sound to make it inaudible. Unfortunately, using this method can destroy the recording. Another method is to use a Pro Tools plug-in that attempts to continue the music before and after the mistake and tries to eliminate any extraneous sounds from the median of the two ends. This method can work for short mistakes.

Observations

2.6 Team Observations

As part of the investigative process for this IQP, the team attended location recordings conducted by area professional recording engineers. By observing experienced sound technicians, proper technique and problem solving were ascertained. Although recording situations were comparable, recording styles often differed greatly from technician to technician.

2.6.1 Recording Session on the West Boylston Common - *Jason Boudreau*

The IQP team first attended an outdoor jazz big-band concert recorded by Jason Boudreau, a WPI student employed by StarFleet Audio in Worcester. The band was amplified through a public address system (PA). It is common knowledge that recording signal directly from a PA's speaker output results in a poor quality recording. This is due to the amount of distortion accompanying mass amplification. Jason strategically placed a spaced pair of microphones to pick up both direct signals from the instruments and the amplified version of the band from the PA speakers. He accomplished this by placing the microphones equidistant from the middle of the band and the PA speaker, and then moved them more to the peripherals or to the center as necessary.

As opposed to many other sound technicians, Jason used only a simple two-channel mixer that only had level adjustment for each microphone. From this mixer, he ran the output into two separate DAT recorders and a separate standard analog cassette for the director's purpose after the performance. His reason for two DATs was to create a backup recording in case any errors occurred in the primary DAT. Jason's levels remained constant except when he lowered levels to prevent clipping, thus capturing a live big band performance with limited editing during the performance.

2.6.2 Recording Session at the Assumption College Chapel - *Nicholas and Bobbie Chase*

Like Jason Boudreau, Nicholas and Bobbie Chase recorded to provide the listener with a firsthand experience. The IQP team attended a recording of the Salisbury Singers performing at the Chapel at Assumption College. Nicholas Chase had set up a spaced microphone pair 15' high, 8' apart and 10' from the chorus. In addition, he also set up a solo microphone, reserved at a zero gain level, for capturing the soloist's sound by properly timing the addition of gain to avoid having the soloist overpowered by the ensemble. The microphone inputs were connected to a level only mixer with the output channeled into a standard Dolby-B Noise Reduction supported analog audiocassette recorder. Nicholas Chase adjusted the levels throughout the recording to deal with the previously mentioned reasons detailed in his interview. On the other hand, Bobbie Chase would not have adjusted the levels unless absolutely necessary.

2.6.3 Recording Session at the Assumption College Chapel - *Dr. Paquette*

An informal observation of Dr. Charles Paquette's recording of the same Salisbury Singers performance demonstrated recording techniques designed for heavy post-production work. Dr. Paquette records mainly for recreation and most are archived in a music library. His styles differ from those of the previous recording technicians because of the sound qualities originally captured during the recorded performances.

During the same choir concert at the Chapel at Assumption College, Dr. Paquette used a high quality omnidirectional AKG stereo microphone raised 14' directly in front of the choir. The internal heads were in the X/Y positions. There was a matching microphone lofted 16' located 40' to stage right, positioned 6" from a large glass wall. This stereo microphone was used to capture the millisecond-delayed echoes off the glass. He stated that this microphone was mixed into auxiliary channels that could later be added to the final recording as an ambient source.

Although he records live performances, Paquette's recordings are post edited to sound like studio recordings. His final recording products lack the location-attributed nuances that the Chases' recordings contain. Dr. Paquette uses heavy postproduction effects to filter out audience noise, performing individual's mistakes and any other undesirable sounds. It takes great skill to execute this aural "airbrushing" technique while retaining the distinct traits of the original performance. Although his style of recording is very pleasurable to listen to, it lacks the "live performance" sound.

2.6.4 **Location Recording Seminar - *Michael Andrews***

On December 3rd 1999, Michael Andrews, a WPI graduate accustomed to equipment similar to that used by the IQP team, held a seminar at WPI about location recording techniques. While attending the seminar, the team absorbed new information regarding some disadvantages in using near-coincident microphone placements.

Andrews explained that while a near-coincident pair represents precise stereo imaging when listened through headphones, its downside involves a hollow sounding centered stereo image when listened to through a typical stereo playback system. It was also learned that arranging a coincident pair would eliminate phase cancellation of high-end frequencies, an anomaly frequently produced by near-coincident arrangements.

After the seminar, the IQP team strongly considered utilizing the coincident pair instead of its near-coincident alternative for the Bancroft Room. However final microphone arrangements would first need to be experimented with and justified with regards to each JOMP ensemble and their respective performance environments.

Chapter 3

Methodology

3.1 **Equipment Analysis**

After extensive literary research and carefully planned interviews, the IQP team prepared to analyze WPI's available equipment.

Available microphones: Two AKG C1000S Cardioid Pattern Condenser Microphones

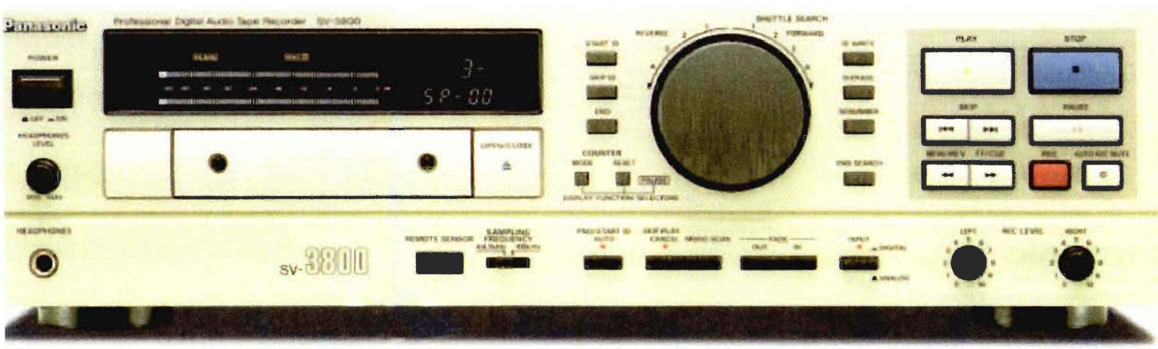
The AKG C1000S is a professional quality microphone. Its highly sensitive frequency response can be adjusted to either of the cardioid or hyper-cardioid pattern settings.

Available mixing console: Mackie Microseries 1202-VLZ Mic/Line mixers

Since condenser microphones require a voltage source, a mixing console with phantom power support was needed. Again, the team's choices were limited to one model, the Mackie Microseries 1202-VLZ. Fortunately, the Mackies are equipped with phantom power capabilities.

Available recording device: Panasonic SV-3800 DAT, Panasonic Analog Cassette Recorder

DAT Back Panel: *Input/Output Console*



DAT Front Panel: *Digital Interface*

This particular DAT model sets the standard for professional digital recording devices. It is capable of all of the digital audio features and "user-friendly" functions (discussed on page 39), and boasts extremely high standards of durability. This model has a most respectable reputation among professionals as its quality is clearly represented by this project's final product. At this point, the IQP team was technically knowledgeable and well equipped for experimentation.

3.2 **Rehearsal Preparation**

After additional research of recording literature was conducted and interviews with audio engineering professionals were concluded, decisions had to be made by the IQP team regarding the recording techniques that would be implemented for this project.

With the general understanding of the art and science of recording at hand, more focused research and applied experimentation techniques were needed to satisfy the project goals. After taking into account limitations and restrictions, including recording conditions and limited available equipment, many recording methods presented themselves as viable options. Since no amount of passive research could produce results as accurate as active experimentation, several tests with available equipment narrowed the prospective techniques into the methods the team eventually used to capture the talent and spirit of the JOMP performing ensembles.

Before any preliminary recordings were arranged with the JOMP ensembles, experiments were conducted focusing mainly on the optimization of the equipment supplied by WPI. Fortunately, the available equipment generally coincided with the recommendations cited in the previous research section. Technical manuals, periodicals and interviews with professional audio engineers had provided the team with a tremendous amount of information, presenting many opinions on optimal techniques. From those resources, the techniques that emerged as superior included the *near-coincident pair microphone technique* for the recordings in the Bancroft Room and the *spaced pair microphone technique* for recordings done in the sanctuary.

3.2.2 **Bancroft Room Performance Preparations**

Observing the characteristics of the Bancroft Room, the environment in which the jazz ensembles were to perform, and interviewing individuals familiar with recording in this room, particularly Nicholas Chase and Bobbie Chase and Richard Ardizzone, revealed many recording restrictions. Jazz recordings previously done in the Bancroft Room have had an undesirable unbalanced reverb that reinforced lower frequencies. Those recordings had muddy characteristics, which were caused by dominating bass reverberation. Commonly, bass reverberation is caused by hard reflective surfaces, which the Bancroft contained. The IQP team decided to simulate an exaggerated bass to discover how to compensate for excessive bass reverberation.

3.2.2.1 Bass Compensation Experiment

In a WPI studio possessing reflective characteristics, similar to the Bancroft Room, the team arranged a small portable stereo system for the playback of familiar jazz music with exaggerated bass tone. The unit was positioned to reflect sound off a hard surface to create a simulated exaggerated bass situation while allowing for a sound source that emitted a broader direct range. No equalization adjustments were performed on the mixer to justify the desired frequency balance, as the purpose of this experiment was to discover microphone configurations and direct source distances that could best capture the reflected sound with the least amount of bass.

It was assumed that because the Bancroft Room's dimensions were considerably larger than the experimental lab, actual optimal microphone distances used during the JOMP jazz ensembles' performances may differ from those detailed in the following experimental data.

Table 1: Bass Compensation Experiment Results

Arrangement	Type	Angle	Distance apart	Distance from source	Results
1	S.S.P*	0°	4'	6'	Muddy sound uncompensated
2	S.S.P.	0°	4'	12'	Similar to above, but weaker signal levels
3	**S.S.P.**	**30°**	**6'**	**6'**	**Cleaner sound**
4	S.S.P.	30°	6'	12'	Clean sound, but with weaker signal levels
5	C.P.*	180°	-	6'	Very muddy sound due to indirect positioning
6	C.P.	180°	-	12'	Very poor sound quality
7	C.P.	120°	-	6'	Slightly cleaner signal
8	C.P.	120°	-	12'	Less clean with increased distance
9	**C.P.**	**90°**	**-**	**6'**	**Very clean sound, moderate stereo spread**
10	C.P.	90°	-	12'	Muddier than previous, poor stereo spread
11	N.C.P.*	160°	6"	6'	Clean sound with moderate stereo image
12	N.C.P.	130°	6"	6'	Better compensation for mid/high range
13	**N.C.P.**	**110°**	**6"**	**6'**	**Very good compensation**
14	N.C.P.	90°	6"	6'	Very good compensation, poor stereo image

*S.S.P. = Spaced Stereo Pair, C.P. = Coincident Stereo Pair, N.C.P. = Near-coincident Stereo Pair

The preceding test results were accomplished by repeatedly recording John Coltrane's "Blue Train" with exaggerated bass using *AKG C1000S* condenser microphones mixed through a Mackie 1202-VLZ onto a Panasonic DAT model *SV-3800*. Many tonal variations were encountered using several predetermined methods of microphone placement, and each recording was later reviewed through the use of the DAT playback feature. While listening to the playback through a high definition headphone set, the team discerned the most favorable synthetic-performance representations.

Three of the fourteen techniques were favored for their reduction in the amount of excessive bass reverb and increasing direct signal. Arrangements 3, 9, and 13 produced clean and undistorted sound captured with enough ambiance to maintain the live performance nuance. Although these three arrangements produced superior results in comparison to the others, all fourteen techniques would once again be tested in full scale in the Bancroft Room during the JOMP student jazz ensemble's rehearsals.

In addition to discovering the significance of microphone positions, the experiment was the team's first complete setup and operation of the recording equipment. The process of setting up the equipment, recording and reviewing gave the team the practice necessary for the production recordings.

3.2.3 Sanctuary Performance Preparation

Orchestral situations on the other hand could not be examined through acoustical simulation, as was used when testing microphone arrangements for the jazz group. The orchestral performances were to take place in the First Unitarian Church sanctuary, a very large room with an approximate 30' ceiling elevation. Long reverberation times are characteristic of environments similar to the proportions of the sanctuary. Since little information was available about the sanctuary's acoustics, further experimentation was needed.

3.2.3.1 Sanctuary Acoustic Analysis

The IQP team conducted tests in the empty First Unitarian Church sanctuary to observe the amount of reverberation present after the original source sound had occurred. Recording quick pulses made from wooden blocks made the degree of reverberation easily observable. As reverb is created by a room's boundaries (walls, floors, and ceiling), microphone angles and spacing remained constant while microphone distances from the ensemble varied throughout the sanctuary.

Table 2: Sanctuary Acoustic Analysis Results

Arrangement	Type	Angle to Center	Angle to Floor	Distance from source	Results
1	S.S.P*	0°	0°	6'	Quick decay
2	S.S.P.	0°	0°	8'	Longer decay
3	**S.S.P.**	**0°**	**0°**	**10'**	**Adequate reverb**
4	S.S.P.	0°	0°	12'	Adequate reverb
5	S.S.P.	0°	0°	18'	Extended reverb
6	S.S.P.	0°	0°	20'	Excessive reverb

*S.S.P. = Spaced Stereo Pair

Resulting from the sanctuary acoustic analysis were two distances that provided favorable results. The team chose the closer 10' distance, versus 12', from the sound source for a slightly reduced reverb because artificial reverb can easily be added, during post-production, to recordings lacking ambience.

3.2.3 **Encountered Preparation Problems**

After testing the equipment's operational status, it became apparent that out of the five *Mackie Microseries 1202-VLZ Mic/Line* mixers available at WPI only three were in proper operating condition for use in these recording projects.

The team also encountered a few problems regarding the WPI Music Department's only DAT recorder. Initially, it was unclear as to why the track memory and location system was inoperable. After quickly referring to the DAT recording manual it was found that the auto-track was engaged, which only assigned new track numbers after 2 seconds of silence. Since this restricted the manual installment of track locations, this feature was clearly undesired for the IQP team's test recordings and remained unutilized during the team's recordings.

Possibly the largest of problems during the recording sessions was finding XLR cables suitable for connecting the microphones to the mixer. The WPI Music Department only had three XLR cables over 20'. With this project requiring two long main cables, and additional spares, procuring these three cables sometimes became difficult. Luckily, all three of WPI's XLR cables were in satisfactory operating condition.

3.3 **Production Recording Preparation** ***(The Rehearsals)***

Using the preliminary tests that were conducted and documented into data log formats, the IQP team began recording JOMP ensemble rehearsals in the production recording environments, namely the Bancroft Room and the First Unitarian Church sanctuary. Preparation for these practice recordings generally started an hour before arriving at the JOMP facilities. The equipment was transported to JOMP by the recording team. The team was required to supply the recording table, DATs, extension cords, splitters, and adhesive tape for securing the cables. The IQP team prepared recording logs (refer to **Appendix A**) with data regarding materials used, techniques to be tested, time of setup, preparations, and sketches of respective ensemble arrangements.

After obtaining all the necessary materials, printing the data log sheets, the equipment was packed into the team's vehicles in personal bags, as opposed to professional carrying cases alone, in an effort to avoid possible theft of WPI owned equipment. Arriving at JOMP well before practice was scheduled, the team's equipment was brought inside the First Unitarian Church and placed inside the respective performance locations.

The purpose of recording the ensemble rehearsals was to familiarize the recording team with the performance environments as well as to familiarize the ensembles with the presence of the recording team.

3.3.1 Jazz Ensemble Rehearsal

During the first of these practice recordings, many additional microphone arrangements were tested along with the original three intended arrangements (refer to Table 1 Results). The additional arrangements were necessary to define levels of reverberation in different locations of the Bancroft Room. Techniques were logged in synchronization with the individual tracks on the DAT recording produced during the sessions. This allowed for the easiest method of recording reviews. Table 3 represents the results of the several microphone arrangements.

Table 3: Student Jazz Rehearsal Experiment

Arrangement	Type	Angle	Distance apart	Distance from source	Results
1	S.S.P*	0°	6'	6'	Poor blend
2	S.S.P.	0°	6'	12'	Very poor blend, excessive reverb
3	C.P.*	30°	-	4'	Clean sound, poor stereo image
4	C.P.	50°	-	4'	Clean sound, better stereo image
5	**C.P.**	**70°**	**-**	**4'**	**Clean sound, desired stereo image**
6	C.P.	90°	-	4'	Excessive drums in left channel
7	C.P.	30°	-	6'	Good sound, monaural image
8	C.P.	50°	-	6'	Clean sound, better stereo image than previous arrangement
9	C.P.	70°	-	6'	Clean sound, prominent drums
10	C.P.	30°	-	8'	Bass reverb appearing, monaural sound
11	N.C.P.*	60°	6"	4'	Clean sound, good stereo image
12	**N.C.P.**	**80°**	**6"**	**3½'**	**Clean sound, desired stereo image**
13	N.C.P.	90°	6"	4'	Prominent drums
14	N.C.P.	30°	6"	6'	Bass reverb occurring from this distance

*S.S.P = Spaced Stereo Pair, C.P. = Coincident Stereo Pair, N.C.P. = Near-coincident Stereo Pair

The team's main focus of attention when reviewing the recorded results included a clear bass tone in compliance with a natural piano effect both accompanied by the desired degree of clarity produced by the drums and horn sections, while maintaining stereo imaging.

After presenting and meticulously reviewing the fourteen experimental recordings with Richard Ardizzone, it was decided that the closely placed (3½' distance from the jazz group) near-coincident pair at the angle of 80° produced the best results. This arrangement coincided with the team's effort to aim the left and right microphones toward the drum kit and piano, respectively. As the drum kit and piano made up the jazz ensemble's left and right peripheral boundaries, distinguishing each boundary instrument in the corresponding left and right channels produced the desired stereo imaging.

3.3.2 **Student Orchestra Rehearsal**

A preliminary recording of JOMP's student orchestra allowed the IQP team to experience the nuances of a performance in the sanctuary environment. During this rehearsal, freedom to move and fine tune microphone locations allowed additional experimentation without the interference of an audience.

Since the orchestra was confined to a narrow area, the members were spread out across the front of the sanctuary between the altar and the pews as opposed to an orchestra's typical semicircular arrangement. The spaced pair arrangement was the only method capable of capturing a stereo representation of such a broad range of sound sources. Since the experimental reverb analysis did not demonstrate the sound projection effects of a full orchestra, additional experimentation of microphone positioning was needed. Using the optimal 8' source distance determined from analyzing table 2, several spaced pair methods were tested with variations in microphone elevation, distance apart from each other, and angles of microphones on all axes. Results of this test developed comparisons between configurations that blended direct signal with desired reverberation.

Table 4: Orchestra Rehearsal Experiment Results

Arrangement	Type	Angle to Center	Angle to Floor	Height	Distance apart	Distance from source	Results
1	S.S.P*	0°	0°	8'	8'	10'	Poor sound quality, very wet reverb
2	S.S.P.	20°	0°	8'	8'	10'	Similar to above, with clearer blend
3	S.S.P.	0°	15°	8'	8'	10'	Similar to #1, crisper high frequency
4	S.S.P.	20°	15°	8'	8'	10'	Clean sound, focus on center
5	**S.S.P.**	**0°**	**0°**	**8'**	**10'**	**10'**	**Good balance, clear sound**
6	S.S.P.	20°	0°	8'	10'	10'	Centrally focus
7	**S.S.P.**	**0°**	**15°**	**8'**	**10'**	**10'**	**Excellent balance, optimal reverb**
8	S.S.P.	20°	15°	8'	10'	10'	Too much focus on center
9	S.S.P.	0°	0°	8'	12'	10'	Focus on far peripherals
10	S.S.P.	20°	0°	8'	12'	10'	Better blend, clearer sound than #9
11	S.S.P.	0°	15°	8'	12'	10'	Loss of balance
12	S.S.P.	20°	15°	8'	12'	10'	Loss of peripheral balance

* S.S.P. = Spaced Stereo Pair

From these results, the team decided which recording arrangements were to be reviewed by Richard Ardizzone. His meticulous comparisons of the presented selections demonstrated his preference for the spaced pair 8' high, 10' across, 0° toward center, and angled 15° toward the floor. This arrangement was used to record the remainder of JOMP's orchestral performances in the sanctuary.

3.4 <u>**Production Recording**</u> ***(The Performances)***

After the microphone placement testing had been completed and techniques were perfected, actual performances and rehearsals were captured using the techniques ascertained from the rehearsal recordings.

The equipment was set up and tested an hour before all performances. An additional analog recorder was used along with the DAT to create a backup copy of the performance. Because the analog recorder was connected to the DAT's output, line levels remained the same for both and none of the recording techniques were compromised. After each performance was recorded, the resulting analog version was presented to Richard Ardizzone for review and selection of individual tracks to be included on the final promotional disk.

3.4.1 ***Faculty Jazz Recital***

December 3rd 1999, 8:00 pm (Bancroft Room)

The faculty jazz recital took place in the Bancroft Room at the First Unitarian Church of Worcester. An audience was present so there were no retakes of any musical piece. The ensemble was arranged as shown below.

Courtesy of David Blondin

Positioned from left to right (on previous page):
Thomson Kneeland - *Bass,* **Mike Connors** - *Drums*, **Richard Falco** - *Guitar,* **Jerry Sabatini** - *Trumpet and Flugelhorn,* **Jim Allard** - *Soprano Saxophone*, *Alto-Saxophone, and Flute,* **Richard Ardizzone** - *Trombone*

3.4.1.1 Microphone Placement and Monitoring Tactics

The microphones were arranged as a coincident pair 3½ feet from horn section, at a height of 7½ feet from the floor. The coincident pair technique was chosen instead of the near-coincident pair to minimize the possibility of high-frequency phase cancellation and hollow sounding stereo imaging, anomalies that are characteristic to near-coincident microphone arrangements. The left channel microphone was angled 15° to 20° down from horizontal and pointed directly between the drums and the upright bass. The right channel microphone was aimed slightly to the right of Richard Falco's guitar amplifier, as it was positioned as much to the right as the left channel horn section microphone was positioned to the left. An internal angle of approximately 80° was maintained. This stereo microphone arrangement preserved the true imaging of the ensemble during playback, with the drums far to the left channel, the trumpet at central channel, and the guitar and trombone far to the right channel.

During Rehearsal: After recording and reviewing the Faculty Jazz Ensemble's first 7:00pm rehearsal piece, the IQP team decided that Richard Falco's amplifier needed to be raised about 1½ feet from floor level. This enabled the guitar to be recorded at the proper input signal level (volume) with respect to the other instruments. It was also apparent that Jim

Allard, who was standing directly between the upright string bass and the left channel microphone, would have blocked the sound from the upright bass. The bassist, Thomson Kneeland, was shifted two feet to his left so that the upright bass's sound would be projected between Jim Allard (saxophone) and Jerry Sabatini (trumpet).

When recording a professional ensemble, it is important to inform the musicians during rehearsal how the recording sounds and to ask them for suggestions and ideas as how to correct the sound. Their insights were very helpful in isolating balancing issues, especially when the problems were due to individual instruments or an overall acoustic anomaly.

The 7:00pm rehearsal also familiarized the team with a few of the Jazz Faculty Ensemble's musical pieces, which helped the team determine the proper level adjustments for the particular performed pieces.

Tracks

Track 1: *Things Change* - Jerry Sabatini

This track was the team's first performance recording. The DAT input signal clipped, (exceeded the digital limit of zero decibels), approximately five minutes and seven seconds (a counter time of 5:07) after the start of the track. The signal was clipped due a loud strike upon Mike Connors' snare drum. This clipped signal is virtually nonexistent to the listener, as snare drums are sometimes meant to be intensely loud. The rest of the track was recorded without incident at the appropriate levels, (about a maximum of -3 decibels).

Track #2: *East of Here* - Jerry Sabatini

The second track was recorded without any noticeable issues.

Track #3: *Serenity* - Jerry Sabatini

This piece was performed slowly and quietly, relative to the other pieces on the program. The Mackie mixer's output gain was adjusted before the piece was performed to appropriately increase the DAT's input levels. Unfortunately, the signal clipped briefly during the middle of the track due to the horn section's increase in volume. The team quickly learned to not adjust levels excessively during a slow musical piece, as sudden volume increase may occur.

Track #4: *Tune formerly known as "Prince Albert"* - Jerry Sabatini

Levels were decreased dramatically after the recording of the previous piece, for this piece was performed at a much higher volume.

Track #5: *You'll Never Know* - Jerry Sabatini

Some fuzz or static was recorded during the middle (around 2:00 after start of track) and also at the end of this piece. This anomaly was thought to be due to moisture in the reed or body of Jim Allard's saxophone, a common nuance heard on many jazz recordings.

Track #6: *Expansion* - Jerry Sabatini

This track was also recorded without any noticeable flaws.

Track #7: *Some Standard* - Jerry Sabatini

The microphones (possibly) detected a distorted squeal toward the end of this piece.

Track #8: *Nica's Dream* - John Coltrane

For this track, the gains were increased, since the levels from the ensemble were inadequate for recording purposes.

Track #9: *Stella by Starlight* – Victor Young

The levels were maintained from the previous track, as the performance volume was low.

Track #10: *Well, You Needn't* - Thelonious Monk

This track contains the same, or at least a similar, effect as heard during track #7. Again, this nuance seemed to be projected from the horn section and was possibly due to moisture contained within the instruments.

Track #11: *Beautiful Love* - Victor Young

This track also contained a static effect, not unlike the previous effect found in tracks 7 & 10. Since this was the final performance of the evening, the team finally determined that this nuance was positively caused by built-up of moisture in the horn section. It was distinctly heard at the beginning and end of the track, upon the entrance and exit of the horn section.

When listening to the sampler selections of the Faculty Jazz Recital performance through a stereo system or headphones, the actual physical arrangement of the group should be clear. It should be evident that the drums, saxophone, and upright bass are supplied more through the left channel, with the drums being farthest left and the bass closer to center. The trumpet's sound should be delivered directly through the center (balanced in both the left and right channels), and both the guitar and trombone are effectively delivered through the right channel with the trombone furthest to the right. Overall, the stereo imaging in the final recording closely resembles the actual arrangement of the performance as displayed.

The encountered technical and acoustic anomalies, previously listed under their respective tracks were evident on the unedited master DAT tapes. The revised prototype disc includes some of the original sound defects.

After the jazz facility concert recordings, the audiocassette was presented to the staff of JOMP for reviewing. The equipment was packed away and returned to WPI. The DAT tapes were then reviewed and archived for later postproduction. All log data forms and coinciding DAT recording information was noted.

3.4.2 ***Student Chamber Trio Production Rehearsal***

Saturday December 4th 1999, 12:00 noon (Bancroft Room)

JOMP's student trio "production rehearsal" was held in the Bancroft Room at noon the day after the faculty jazz recital and was the team's second production recording. Mr. Ardizzone decided that this student trio, along with the other student ensembles scheduled for production recording, was more comfortable being recorded during a well-rehearsed production rehearsal as opposed to an actual performance presented to an audience. This particular trio consisted of a flautist, a cellist, and a pianist.

The student trio was arranged from left to right as in the following representation.

Courtesy of David Blondin

Positioned from left to right:

Paul Wright - *Cello*, **Liana Popkin** - *Flute*, **Keithe Baggett** - *Piano*

3.4.2.1 Microphone Placement and Monitoring Tactics

Because this rehearsal was held in the Bancroft Room, the microphone setup differed only slightly in comparison to the arrangement used during the faculty jazz recital. To achieve the desired microphone arrangement, one team member monitored the blend of the trio's warm-up session via *Seinheiser* headphones while the other manipulated the microphones to the first's liking. This process was repeated a few times as the team's members alternated roles of monitoring and positioning until the desired blend and imaging was agreed upon. The microphones were finally arranged as a coincident pair, changed from the initial intention of using a near-coincident pair as to ensure minimal high-frequency phase cancellation. The microphones were positioned 3' from the trio at an elevation of 7'. As opposed to angling the microphones equally toward the floor and equally from each other, the recording team decided to aim the left channel microphone between the sound fields of cello and the flute, although more toward the cello to maintain the cello as the left channel image boundary, and the right channel microphone directly toward the center of the bottom edge of the closed piano. (When a piano is closed, most of its sound resonates from the bottom of the instrument.) Although unique, this process of microphone positioning proved effective during the team's never-before encountered trio production recording and is further justified through stereo playback. As there was no audience present, the trio was allowed to repeatedly attempt to play the movement of the composition intended for the compact disc production.

Student Chamber Trio Tracks

<u>Track 1: *Allegro -con brio*, Trio in Bb Op. 11 Beethoven</u> (1st attempt)

The first attempt of this piece was performed in its entirety, although it did not seem to satisfy the liking of Mr. Ardizzone. Upon calling his wife, Wendy Ardizzone, who was at that moment off location, Mr. Ardizzone advised both the trio and the recording team to wait until Mrs. Ardizzone arrived to instruct the remainder of the trio's attempts.

Upon reviewing the first attempt using the DAT's playback feature, Mrs. Ardizzone instructed the students to perform the second attempt at a slower tempo, as she suggested that the first attempt might have been performed too fast.

<u>Track 2: *Allegro -con brio*, Trio in Bb Op. 11 Beethoven</u> (2nd attempt)

Performed in its entirety and considerably slower than the first, this 2nd attempt appeared to meet the Ardizzones' high expectations. However, possessing the meticulous ears of a musician, Mrs. Ardizzone suggested that the students' performance might improve by rehearsing only a portion of *Allegro* as opposed to the entire piece.

<u>Track 3: *Allegro -con brio*, Trio in Bb Op. 11 Beethoven</u> (3rd attempt)

This 3rd and shortened attempt, failed to achieve the Ardizzones' expectations and one more attempt was suggested.

<u>Track 4: *Allegro -con brio*, Trio in Bb Op. 11 Beethoven</u> (4th attempt)

This 4th attempt, although somewhat better than the third, matched only the quality of the second and was the final recorded performance of the student chamber trio.

As the recording session drew to a close, the team presented the Ardizzones with the duplicate, analog cassette version of the trio's recordings. After reviewing the cassette, the Ardizzones decided to use the trio's first full attempt, track 1, of Beethoven's *Allegro - con brio* for the final promotional disc.

When listening to the student trio's performance of *Allegro – con brio*, careful attention should be paid to the stereo imaging. Although all instruments were received through both microphones, the cello creates the left channel boundary and the piano creates the right channel boundary as planned. The flute is represented through the center channel, or equally through both channels, and is imaged somewhat further away than the cello and the piano. It is clear that the overall imaging of this recording is a true aural illustration of the trio's physical arrangement.

3.4.3 ***Student Orchestra Production Rehearsal***

December 6th 1999, 7:30pm (Sanctuary)

The orchestra, directed by Timothy Terranella, consisted of eighteen members in six sections. The sections included violins, violas, cellos, double bass, and flutes.

Picture of Orchestra Arrangement *Courtesy of David Blondin*

Similar to the previously mentioned conditions of the student chamber trio recordings, the orchestra was also recorded during a rehearsal as opposed to a performance in order to eliminate the interference from an audience and to allow multiple recordings of the same piece if errors occurred. Although this initially seemed prudent, the orchestra, as apposed to smaller ensembles, lacked the mental preparation usually present during a live performance. This may have reduced the performance quality of the individual musicians and, consequently, the orchestra as a whole.

Before the rehearsal, the recording team was informed that a harpsichord would be accompanying the orchestra during a few selections. The harpsichord is generally a quiet instrument and is sometimes difficult to record when accompanying an ensemble the size of an orchestra; such was the case with JOMP's student orchestra. The IQP team was prepared with a spare *ElectroVoice* omni-directional, dynamic microphone and extra XLR cables.

As mentioned, the orchestra was tightly arranged along the front of the sanctuary between the altar and front pews, so once the orchestra had set up, there was no room in front of the orchestra where harpsichords are usually located due to their dwarfed sound. The harpsichord had to be located to the far right of the orchestra, next to double bass shown in the picture on the previous page.

3.4.3.1 Microphone Placement and Monitoring Tactics

The two main stereo microphones were located 8' high, 8' from the orchestra, 10' apart in the fourth pew, predetermined by the research and experimentation. The microphone capsules were angled 15° toward the floor and aimed directly forward. This arrangement provided the team's preferred blend of direct and reverb sound.

Through impromptu acoustical analysis, the team decided that the third, ElectroVoice dynamic, microphone was to be placed approximately 3' from the harpsichord, which faced into the rear wall of the altar. Aiming directly at the center of the harpsichord's opened resonance board and elevated 3½ feet from ground level (see picture of harpsichord below), the dynamic microphone captured the harpsichord sound that the original left and right channels lacked. In accordance, a separate cable was taped down and strung from the third microphone to the far left staircase of the sanctuary, where the team's members and their equipment were positioned to avoid distracting the musicians.

Depiction of the harpsichord microphone *Courtesy of David Blondin*

The harpsichord's microphone channel was panned slightly off center toward the right channel to give the harpsichord the effect of being in front of the orchestra, while maintaining the actual imaging of its presence to the right.

Being a rehearsal, the orchestra frequently stopped, reviewed sections of music, and started again. Having recorded many fragmented movements made post-production difficult, as each take had to be reviewed for completeness as well as quality.

Orchestra Session Tracks

<u>Tracks 1 through 5: *Joc cu Bata* - Bartok</u>

These five tracks are made of three incomplete attempts at the movement and two complete recordings.

<u>Tracks 6 through 11: *Braul* and *Pe Loc* - Bartok</u>

These tracks contain fragments and two complete recordings of two seamless movements performed without a clear division between the two. They needed to be distinguished in the post-production.

<u>Tracks 12 through 20: *Buciumeana* and *Poarga Romanbasca*</u>

These eight tracks are much like the previous, as there are indistinguishable pauses between the movements. These, as well, needed to be separated in post-production.

<u>Tracks 21 through 30: *Maruntel*</u>

These last recordings contained many segments with only three complete takes.

<u>Tracks 31 through 35: *Concerto Grosso -Overture* - Handel</u>

This movement was many fragments with one complete track.

Track 36: *Concerto Grosso -Allegro*

Complete movement. The conductor seemed quite pleased with this take.

When listening to JOMP's orchestral production rehearsal recordings, it is apparent that the stereo image position of the harpsichord is correct. Slightly to the right of center, the line input level of the harpsichord's microphone was balanced so that its volume intensity did not appear too great or too insignificant. Essential harpsichord reverb is also present due to the *ElectroVoice*'s omni-directional capability of capturing many reflected sounds. The importance of the choice of this microphone should not be underestimated.

3.4.4 ***Faculty Chamber Trio Production Rehearsal***

December 6th 1999, 9:00pm (sanctuary)

Immediately after recording the student orchestra, the team recorded a baroque style trio consisting of an alto recorder, a harpsichord, and flute. Since there were only three musicians, they were spaciously arranged in the middle of the sanctuary's altar area. The harpsichord was repositioned so the sound would project outwards into the sanctuary. The recorder and flute faced each other to maintain visual contact for musical cues. Refer to photo on next page.

Courtesy of David Blondin

Shown from left to right:

Marina Minkin - *harpsichord*, **Tim Terranella** - *flute*, **Jerry Bellows** - *alto recorder*

Informed of this recording only a few days before it actually took place, the team was unfamiliar with the trio's physical arrangement and musical style. Again, the team's exhaustive research aided in an outstanding improvised microphone arrangement.

3.4.4.1 Microphone Placement and Monitoring Tactics

The team decided to leave the microphones in the same location, but rotate each head 20° toward the altar. Essentially, they were aimed at the harpsichord with the wind instruments intersecting the cardioid path. Since the musicians would be facing each other, their sounds would overlap laterally while the harpsichord's sound would project forward through them. This imaging technique proved to be successful and is justified by the remarkable results of the recording.

The third microphone used with the harpsichord during the orchestral recording was set up, as observed in the picture. This microphone was not used, although the recording team attempted to strengthen the harpsichord's sound. Because the microphone configuration was based on exhaustive research and experience, the third microphone was unnecessary because the stereo pair was well focused on the harpsichord.

Faculty Chamber Trio Tracks

Tracks 1 and 2: *Trio Sonata in C -Affettuoso* - Quantz

These two practice takes allowed the recording team to manipulate and monitor several subtle microphone variations.

Track 3: *Trio Sonata in C -Affettuoso* - Quantz

The third track was a complete movement.

Track 4: *Trio Sonata in C -Larghetto* - Quantz

This selection was a complete take, but contained subtle performance blemishes.

Track 5: *Trio Sonata in C -Larghetto* - Quantz

This second attempt was more admirable than the first containing few errors.

Track 6: *Trio Sonata in C -Vivace* - Quantz

The final track was a single take with very few errors.

The team's successful capture resulted in the brilliance of the woodwind instruments accompanied by the surrounding ambiance of the harpsichord. Flautist Tim

Terranella exclaimed, "It's the perfect blend!" after reviewing a few of the recorded selections through the DAT's playback feature.

3.4.5 ***Student Jazz Ensemble Production Rehearsal***

Wednesday December 8th 1999, 7:00pm (Bancroft Room)

JOMP's student jazz ensemble marked the last of the recording sessions. The ensemble was comprised of six members and coached by JOMP assistant director, Richard Ardizzone. The six members included a pianist, an upright bass player, three horn players, and a drummer.

Courtesy of David Blondin

Arranged from left to right:

Mark Zaleski - *soprano sax,* **Stephen Schwall** - *drums*, **Ben Glasser** - alto sax, **Joshua Filgate** - *trombone*, **Max Zeugner** - *double bass*, **Corey Bernhard** - *piano*

3.4.5.1 Microphone Placement and Monitoring Tactics

As well as having learned much throughout the previous week's recording sessions, the team already had experience in recording this group from the first experimental recording in the Bancroft Room. Analyses of the previous recordings led the team to believe that a coincident pair, similar to that used for the faculty jazz recital, arranged 3½' from the sound source, internally angled at approximately 80°, and elevated 7½' would provide the best results. As with the faculty jazz ensemble, the microphones were aimed at the appropriate stereo image boundaries. The left microphone was aimed at the drum kit and the right microphone was aimed at the piano, which was opened approximately 6" to create a volume balance with the other instruments.

From experience gained during previous recordings, the recording team learned that the student jazz drummer was somewhat louder in comparison to the drummer of the professional jazz group. With this in mind, three pieces of available plush furniture were used to surround the drum kit with piano covers covering the gaps. This technique dampened the sound, which added clarity by reducing the amount of reverb that would have occurred from booming lower toned drums. During preliminary recordings for levels, it was discovered that the ride cymbal, being located above the couches, was unaffected by the dampening effect. To counteract its extreme brilliance, two music stands were elevated 5' from the floor and positioned to absorb the direct sound from the cymbal crown to the left channel's microphone. A plush jacket liner was draped on these music stands, which absorbed excessive high frequency sounds. This sound-dampening technique is depicted on the following page

Courtesy of David Blondin

During the actual recordings, the group had to start over several times for each piece. This made recording difficult, as the team was uncertain when to pause the recording and when to continue. The director, Richard Ardizzone, asked the recording team to perform difficult techniques (such as overdubbing) during the recording session. This is very uncommon in location recording and the team was not prepared for such delicate and arduous procedures. Although three failed attempts were made to overdub, a drum break in the musical piece allowed for clean digital post-production sound editing since the drum solos for this piece contained moments of silence.

Student Jazz Ensemble Tracks

Track 1: *Things Change* – Jerry Sabatini

This first track was intended to be a practice, but became the band's best performance of this piece. This is most likely because the band was not as nervous as when acknowledging that they were being recorded.

Tracks 2 through 4: *Things Change* – Jerry Sabatini

These tracks involved incomplete takes of the song punctuated by Ardizzone cutting the band during the piece.

Track 3: *Things Change* – Jerry Sabatini

This track is the ensemble attempting to perform the piece in its entirety. The band did not perform as well as before, as they were preparing for the recording and became nervous.

Track 8: *Tune Formerly Known as "Prince Albert"* – Jerry Sabatini

This is the first of three takes of this piece already recorded by the IQP team during the professional ensemble.

Tracks 9-10: *Tune Formerly Known as "Prince Albert"* – Jerry Sabatini

These two attempts at recording this challenging piece did not capture the ensemble at their finest.

Track 12: *Samba de los Gatos* – Mike Steinel

This piece, intended to be another warm-up piece eventually became one of the group's better performances, as they had been practicing the piece longer than the others.

Track 17: *Midnight Mambo* – Oscar Hernandez

This first attempt at performing this piece, the ensemble had difficulties, as they had only recently started practicing this piece.

Track 19: *Midnight Mambo* – Oscar Hernandez

This take became the best out the three complete takes.

Track 20: *Midnight Mambo* – Oscar Hernandez

The drummer was getting too excited about this piece and started playing too loudly.

When listening to the recordings of the student jazz ensemble, notice the decrease in the percussionist's volume in comparison to that of the faculty jazz sextet percussionist's. The recording team's makeshift drum dampening system quieted the volume of the drums considerably. Although the excessive sounds from the cymbals, snare, and toms were lowered as desired, the kick drum's sound may have been dampened a bit too much. After listening to the master DAT recording of the student jazz ensemble, the team decided that opening a gap between the couches and no higher than 3' from the floor would probably have allowed for a desired kick drum volume.

3.5 Post-production

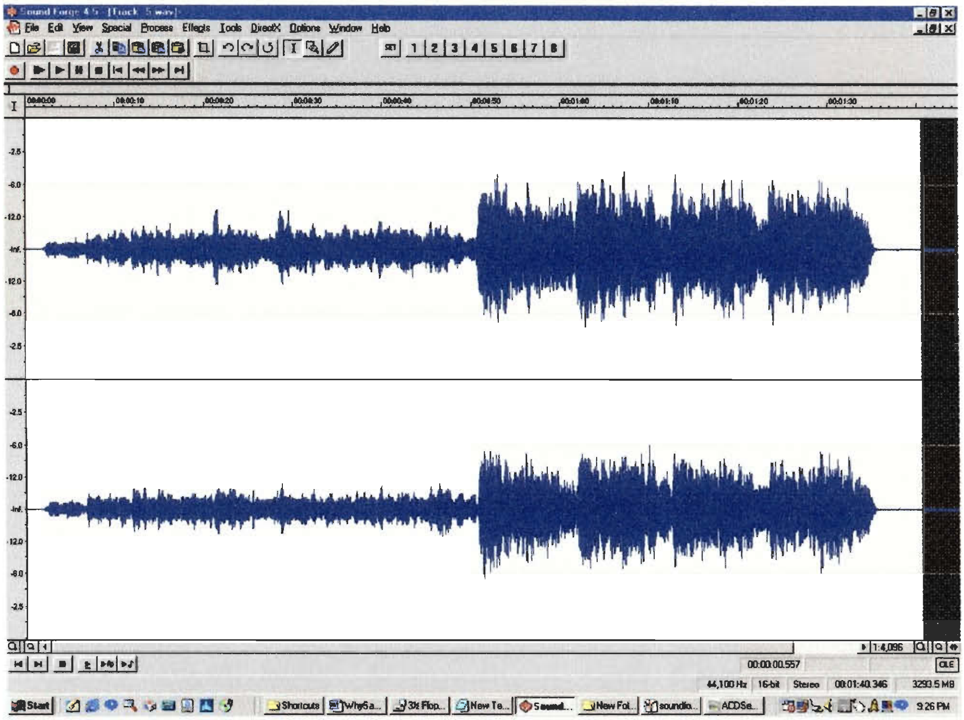
3.5.1 Sound Editing and Audio Format Transfer

After all the performances had been recorded, the next step was to transfer the DATs to a computer file for editing. This process became the major stumbling block for the IQP team. At the beginning of the project, the team was told that the WPI Music Department had the capabilities of transferring the DATs to various formats using ProTools, the most popular Macintosh-based audio editing tool. From extensive research, the team determined the individual tracks needed to be converted to the IBM/Macintosh compatible *wav* file. This form of file is easily manipulated using both simple and sophisticated sound editing programs on both platforms. From simple conversion, the *wav* file can be converted to the 44.1kHz Pulse Code Modulated (PCM) data necessary for transfer to standard audio CD. In addition, *wav* files have the most accurate recording capabilities for digital-to-digital transfer. Consuming around 60MB (megabyte) for an average song, *wav* files require enormous storage mediums. The only economical means of storage was a standard data/audio compact disk, which are inexpensive, easily transportable, and platform independent.

After reviewing the manuals for the hardware interface and the software guides, the team attempted to use WPI's custom Macintosh G3 designed specifically for audio/movie editing. After many failed attempts to activate the interface between the hardware and the software, the team finally decided that the Music Department's equipment was no longer an option for the transfer.

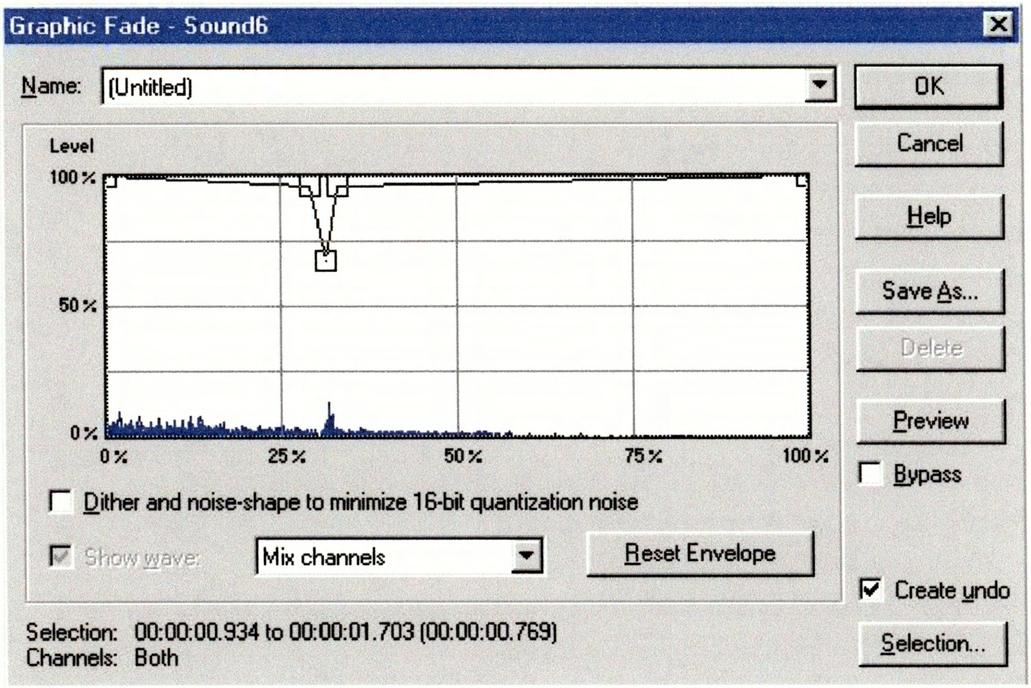
After additional research of the more familiar IBM style platform-based hardware/software, the team discovered that digital transfer capability was possible using certain soundcards such as CreativeLabs' series of *SoundBlaster Live!* and Voyetra Turtle Beach's *Montego* soundcards. These cards boast S/PDIF interfaces for stereo digital input, which can be used by many sound-editing programs, which was exactly what the IQP team needed. Once the transfer was completed, the files would need to be transferred to a CD, since CD writers are commonly found in high-end computer systems as well. Since many high quality computers have either an aforementioned or comparable soundcard, finding individuals at WPI with these systems was not difficult. The difficulties arose when trying to allocate time to use these systems, as the transfer process is arduous, taking thirty minutes to transfer an average song. Additionally, post-editing work should ideally be done on the same computer system on which the files were transferred for convenience and consistency. Since the students at WPI find it very difficult to part with their computer for the duration that the team required, the team found it nearly impossible to find systems capable of the work necessary for a quality final product.

After many futile attempts to the transfer using the prescribed methods, the team had no reasonable choice. Because the deadline for submission was approaching, the IQP team decided professional help was necessary. Jason Boudreau, who had helped the team earlier in the project, was contacted through Star Fleet Audio, a local audio production studio. From the track listing, the individual tracks were dubbed to Sound Designer II file format. Because this format is a standard audio format, which the team could use via the editing program Sound Forge 4.5, the team did much of the post-production on a personal computer. The files were compressed to create a uniform volume throughout the individual tracks. Some digital reverb was added to the student trio, and to both the student and professional jazz ensembles. Having a minimal amount of reverb as planned by the team proved beneficial in this process, as reverb can be added, but usually cannot be removed from a recorded piece.



3.5.1.1 Example of Sound Forge 4.5

Each track was cleaned with noise gates, a process that removes extraneous noise during periods of silence, and faded in and out to prevent the sudden popping sound frequently occurring as a speaker receives a sudden peak in amplitude. The most dramatic editing that needed to be done was the cleaning of the beginnings and ends of each track. Because the students seemed to consistently forget that they were being recorded, their actions were unrestrained. Moments after a piece ended, the musicians carelessly dropped sheet music, coughed, and on one track, a student started talking. Since the effects of reverb are sustained after the sound source ceases, the decay is a characteristic necessary for consistency. The extraneous sounds could have ruined the recordings, but through careful digital manipulation in Sound Forge, the sounds were reduced. By creating a custom fade pattern containing a sudden notch in volume intercepting the noise, the unwanted sound was dramatically reduced (refer to picture below).



To replace the present, but subtle, decay immediately after the music stopped, a millisecond sample of the sound directly before and after were extended into the gap. The fade was then manipulated again with a small spike to restore the removed sample and the last few seconds of the piece was processed with minimal reverb to blend any imperfections. Although this technique is very uncommon in the professional recording industry, the team perfected the process through experimentation to create a seamless edit.

3.5.2 **Graphics Compilation**

As the team finalized sound editing and format transfer, David Blondin printed CD graphic/text liners using high quality printers, surpassing the capabilities of the available equipment owned by WPI. The team's graphics work focused mainly on labeling the sampler disc, itself. The team decided to use the WPI IMC Lab's CD printer to print an image and some text onto the top, unreadable side, of the sampler disc. This process was chosen over its alternative labeling technique, *stomping*. Stomping involves the placement of an image and text written sticker onto the top of the disc. Where this simple technique costs much less to process, its downside involves a tendency to create an unbalanced spin on the disc when used in some CD players. The trade off between playability and cost, here, was too great of risk, so the team printed the label.

(See **Appendix E** for the sampler graphic templates and **Appendix F** for the sampler text templates)

Because the Ardizzones desired an action photograph of Jerry Sabatini placed underneath a clear CD tray, the team needed to find specialty jewel cases being sold in reasonable quantities. Cases are usually sold in bulk quantities, usually in multiples of 50 or by the gross. Although, the company working through http://www.XDR2.com allowed purchases of the special jewel cases in flexible quantities, which prompted the team to purchase twenty cases through them. Once the liners, labels, and CD had been produced, fifteen CD's were distributed after they were assembled.

(See **Appendix B** for detailed graphics discussions)

After this project was completed, the team reviewed its progress and all of its difficulties. Several recommendations to facilitate future recording projects can be made.

3.6 Conclusions

Commonly, professionals record directly to a computer to reduce the amount of steps necessary to produce a CD. PC based computers can be customized for recording relatively inexpensively, as the price of computer components are fairly low. Since a computer used for recording requires accuracy with high-speed sampling, the machine should be made with either SCSI devices or comparable quick-access hard drives. Since computers being sold today are designed to run very quickly, purchasing a system with a fast microprocessor and high-speed devices is inevitable. Included with these packages is a large amount of RAM, which allows large amounts of temporary storage for recording and post-production. The most important additional component necessary is the soundcard.

Consumer level soundcards are usually acceptable for recording voice for entertainment. Professional level recording soundcards are designed with high quality microchips including the digital signal processor and the analog to digital converter. These components are sensitive to electronic interference, which is always present within a computer. Several soundcards are isolated from the computer by having the microchips outside the machine in a shielded enclosure with only the digital inputs being fed into the computer. This technique produces the best results, as the producers of this equipment are designing quality products intended for professional level use. The price for an external soundcard usually starts around $200 for the essential digital interfacing. More advanced devices include multi-track recording and various interfaces including fiber optics, analog input/output, MIDI connectors and sophisticated software.

In addition, the computer must be equipped with a CD writer to allow the production of a CD after post-production. Having a computer equipped with the mentioned hardware would have shortened the CD production by a complete term, as the team attempted transfers from the DATs using poor quality soundcards and confusing Macintosh software.

Finally, any future team should completely research the capabilities of WPI services. Some branches, including the IMC, the Graphics Lab, and the Communications Group have resources far beyond what a team can accomplish alone.

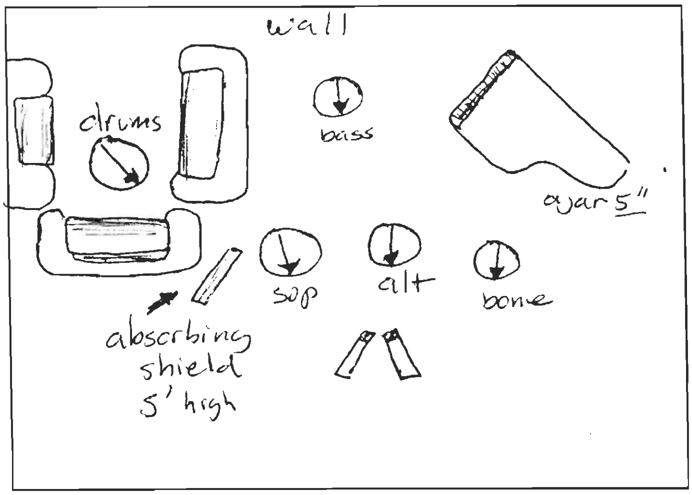
Chapter 4

Appendices

Appendix A

Recording Data Log
Group: Student Jazz Ens. Date: 12-8-99 Time: 8:00
Location: Bancroft

General Room-Band-Microphone positions



Stereo Microphone Placement

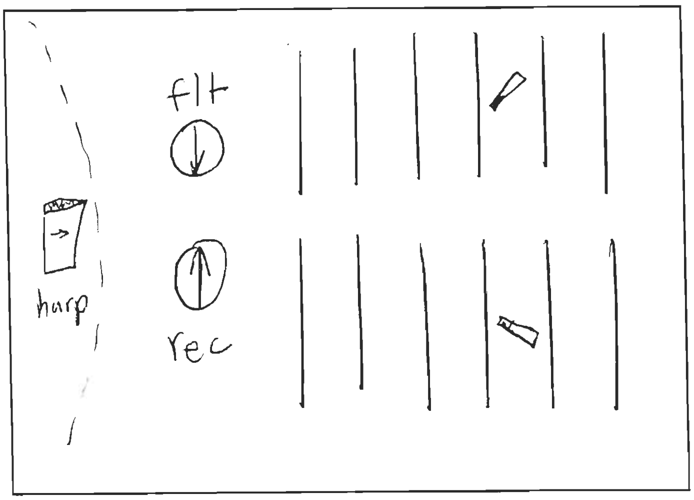
Method: coincident
X/V-Angle: 80°
Grill Distance: –
Source Distance: 3 ½'
Height: 7½'
Distance Apart: –
Angle Down: 10°
Angle In: –

Track	Title	Start	End
1	things Change	00:00	2:50
5	"	3:40	6:00
8	tune Formerly	15:39	17:17
9	tune formerly	17:17	18:10
10	tune formerly	18:10	26:59
11	practice	25:59	26:00
13	"	27:54	29:12
16	"	35:22	38:31
17	Mambo	39:39	41:50
19	Mambo	41:50	42:50
20	Mambo	42:00	∞

the drummer was too loud - sofas & jackets were used to block direct sound

Recording Data Log
Group: Prof. Baroque Trio Date: 12-6-99 Time: 8:00
Location: Sanctuary

General Room-Band-Microphone positions



Stereo Microphone Placement

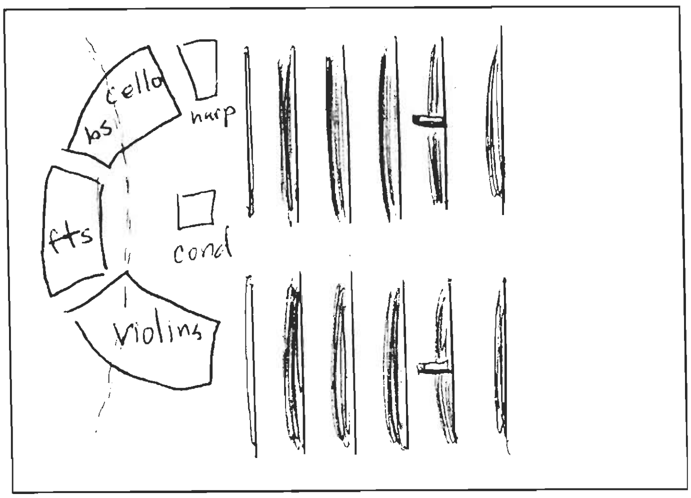
Method spaced pair
X/V-Angle –
Grill Distance 10'
Source Distance 8'
Height 8'
Distance Apart 10'
Angle Down 10°
Angle In 30°

Track	Title	Start	End
1	Trio Son. Affett	02:23	04:20
2	"	04:40	06:21
3	"	07:23	09:53
4	Trio Son Larg	10:21	13:23
5	"	14:53	16:42
6	Trio Son Vivace	17:00	19:42

It was assumed the harpsichord would again need a separate microphone, but it was not used.

Recording Data Log
Group: Student Orchestra Date: 12-6-99 Time: 6:00
Location: Sanctuary

General Room-Band-Microphone positions



Stereo Microphone Placement

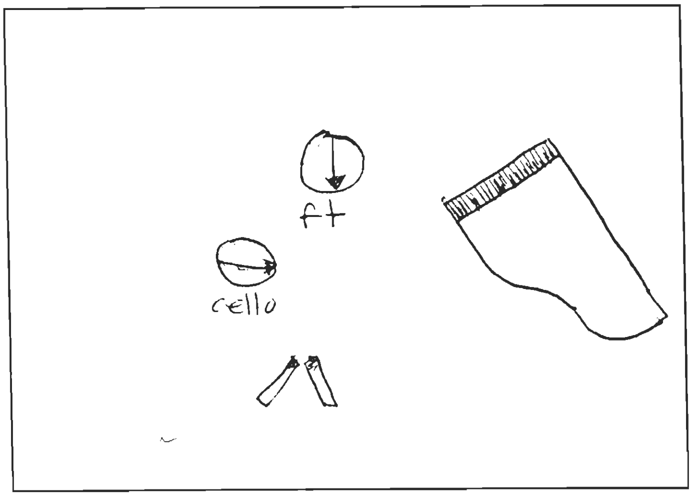
Method spaced pair
X/V-Angle 0°
Grill Distance 10'
Source Distance 8'
Height 8'
Distance Apart 10'
Angle Down 15°
Angle In 0°

Track	Title	Start	End
1-5	Joc cu Bata	2:35	15:36
6-11	Brául	17:43	23:46
12-20	Bucumencinci	25:11	30:21
21-30	Maruntel	32:00	40:52
31	Concerto Grosso	52:21	59:32
35	Concerto Grosso	1:01.01	1:03:08
36	Con Gr. - Allegro	1:07:21	1:04:32

The addition of an harpsichord warrented the use of another microphone

Recording Data Log
Group: Student Trio Date: Dec 4, 99 Time: 10:00
Location: Bancroft

General Room-Band-Microphone positions



Stereo Microphone Placement

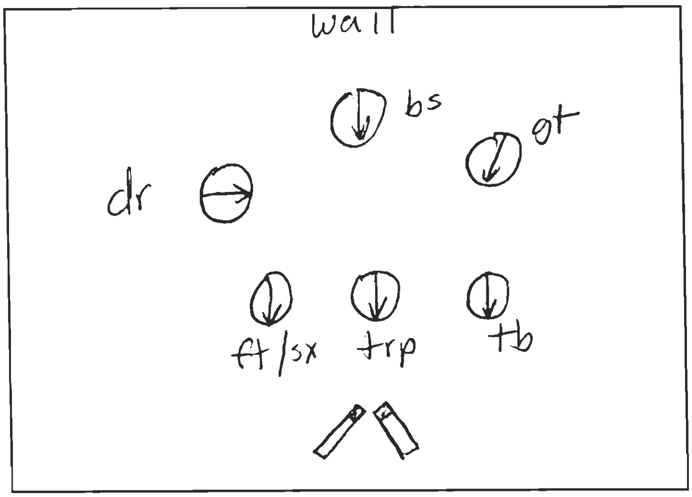
Method coincident
X/V-Angle 50°
Grill Distance .5"
Source Distance 3'
Height 7'
Distance Apart
Angle Down
Angle In

Track 1	Title Trio IV Opus XI	Start 00:00	End 3:40
Track 1	Title Opus XI 1st brio	Start 3:41	End 5:56
Track 2	Title Opus XI 1st	Start 6:00	End 8:23
Track 3	Title "	Start 11:00	End 13:22
Track 4	Title "	Start	End
Track	Title	Start	End
Track	Title	Start	End
Track	Title	Start	End
Track	Title	Start	End
Track	Title	Start	End
Track	Title	Start	End

Piano remained closed because of volume imbalances

Recording Data Log
Group: Professional Jazz Date: Dec 3rd 99 Time: 800
Location: Bancroft

General Room-Band-Microphone positions



Stereo Microphone Placement

Method: Coincident
X/V-Angle: 60°
Grill Distance: .5"
Source Distance: 4'
Height: 8'
Distance Apart:
Angle Down: 10°
Angle In:

Track	Title	Start	End
1	Things Change	0:53	7:25
2	East of Here	7:25	16:40
3	Serenity	16:40	22:50
4	Tune Formerly..	22:50	33:00
5	You'll Never Know	33:00	40:00
6	Expansion	40:00	47:00
7	Some Standard	47:00	56.00
8	Nica's Dream	56:00	60:08
9	Stella by Starlight	1:08	—— new tape
10	Well You Needent	00:00	06:01
11	Beautifl Love	0:601	∞

the drummer faced the horns, as opposed to facing the audience

4.2 <u>Appendix B:</u> Discussions Concerning Sampler Graphics

4.2.1 Graphics Discussion with Marguerite Isaacson and Eleanor McCrea

November 16th 1999

4.2.1.1 *Initial Plan of JOMP's Sampler Compact Disc's Encasement Elements*

- professional standard single CD plastic encasement w/ transparent "disc-bed"
- single sheet front cover consisting of graphics and text on its front and an image with text and production credits on its reverse
- single sheet back cover consisting of graphics and text on the outside with photo image on its reverse side (inside, behind transparent disc-bed)
- graphics were produced by David Blondin
- text compiled by Richard Ardizzone
- prototype sampler CD text and graphics arrangements completed by Marguerite Isaacson and Eleanor McCrea.

In keeping with the timeline for the CD sampler graphics production, the IQP team presented the WPI Communications Group with a mock cover-graphics sheet. The four templates included the CD's front cover, front cover reverse side, back cover, and the image behind the transparent disc-bed on the reverse side of the back cover (See page 108).

Upon speaking with Marguerite Isaacson, the Communications Group graphic designer, and Eleanor Mccrea, the Communications Group production manager, the team decided that David Blondin, having extensive skills in photography and graphics manipulation, was well equipped to create the graphics for all four of the respective disc encasement components. In compliance, a copy of the mock cover-graphics templates was handed to David Blondin. Later, in January, Blondin presented Richard Ardizzone with a portfolio of prospective CD cover art, which included his own representative style and font for the front cover image and text. Mr. Ardizzone considered Blondin's suggestive ideas and later agreed that his designs assisted in capturing the attention of prospective sampler disc consumers.

4.2.2 **Initial Meeting with A Graphic Artist *(David Blondin Jr.)***
November 17th 1999, 5:20 p.m.

Upon arriving at the Joy of Music Program David Blondin, local photographer and graphic artist, briefly introduces himself to the sampler production director, Richard Ardizzone.

Mr. Blondin first asked Mr. Ardizzone exactly what he wanted to see in the JOMP sampler's cover-graphics. After showing Mr. Blondin both performances halls, the Bancroft Room and the First Unitarian Church sanctuary, Mr. Arizzone expressed how concerned he was about the low-lighting situation in each of the two rooms and immediately suggested that a group photo be taken of each ensemble. He then explained how each performance group was expected to show up approximately one hour before its performance and, therefore, Mr. Blondin would have sufficient time to photograph each group at that time. Blondin agreed to this idea and mentioned that it would also give him the opportunity to use a photographer's flash, if necessary, without distracting the ensemble or audience members during the performances. Ardizzone then told Blondin that he could place the respective ensembles in any nearby setting, particularly suggesting that photos be taken set against the background of the spiral-staircase in the sanctuary and fireplace in the

Bancroft Room. Ardizzone further mentioned that placing the Faculty Jazz Group in front of the Bancroft Room's lit fireplace on the night of it's performance, December 3rd 1999, was suitable since that particular ensemble consisted of only six members. He decided that the Faculty Jazz Group would pose with their instruments next to the Bancroft Room's lit fireplace, with an equal amount of ensemble members on either side. The other larger ensembles were to pose in the sanctuary in front of its alter or the aforementioned spiral staircase. None of the group pictures were actually arranged, as Mr. Ardizzone informed Mr. Blondin and the IQP team that action photographs, where students were captured during performance, were desired.

Mr. Blondin, noting everything Ardizzone suggested, then proceeded to describe his operation during the photo shoots, that he would bring three rolls of film, one color, one black and white and one special roll of high-speed film, and two cameras if necessary to capture a variety of photographs. The high-speed film, Blondin mentioned, was appropriate for capturing the essence of the performances in their natural, low-light environments.

4.2.3 **Initial meeting with Mr. and Mrs. Ardizzone**

January 8th 2000

Upon reviewing David Blondin's photographs of JOMP's production rehearsals and performances, the team presented Richard Ardizzone with all photographs along with a portfolio of the most promising potential CD photographs.

From eight rolls of film one was black-and-white and seven were high-speed (flashless) color. Eight packets containing twenty-four photographs each produced an approximate total of one hundred and sixty-eight color and twenty-four black-and-white photographs of the recorded JOMP ensembles.

Mr. Ardizzone was impressed by the broad selection and quickly decided that the initially purposed four CD graphic templates be increased to six templates. The front cover, which the team originally planned on restricting to only two photographs (one photo for the front and one for the reverse side), was asked by Ardizzone to be made of a single folded piece of paper with one picture on each of the four consequent templates.

The two inside templates consist of faded photos overlaid with black representative text. The front cover photograph is not faded but contains the overlaid CD title in bold, white letters. The folded cover's fourth and final photograph is found opposite of the disc itself upon opening the CD jewel's case. The remaining two photographs are printed on opposing sides of a single sheet of paper making up the rear panel of the jewel case. See pages 107-112)

With the IQP deadline (March 2nd 2000) approaching, the team decided that contracting two separate affiliates, Mr. Blondin and the Communications Group, to work on one graphics production was simply too big of a risk. The team's project advisor, Professor Richard Falco, informed the team that the Communications Group might not be able to finalize the CD graphics before the project deadline. And so, as David Blondin had appropriate graphic editing skills and was already familiar with all of JOMP's performance and rehearsal photographs, the team decided not to involve the WPI Communications Group with the final production.

4.2.4 **Final meeting with David Blondin, and Mr. and Mrs. Ardizzone**

January 15th 2000

During this meeting David Blondin accompanied the team in presenting Wendy Ardizzone and Richard Ardizzone with many photo templates, properly cropped, sized, and printed to fit a jewel case. These templates consisted of the photographs that Mr. and Mrs. Ardizzone selected from the initial CD graphics meeting, and a few extra photographs that Mr. Blondin had chosen because of how well they fit the cropping boundaries.

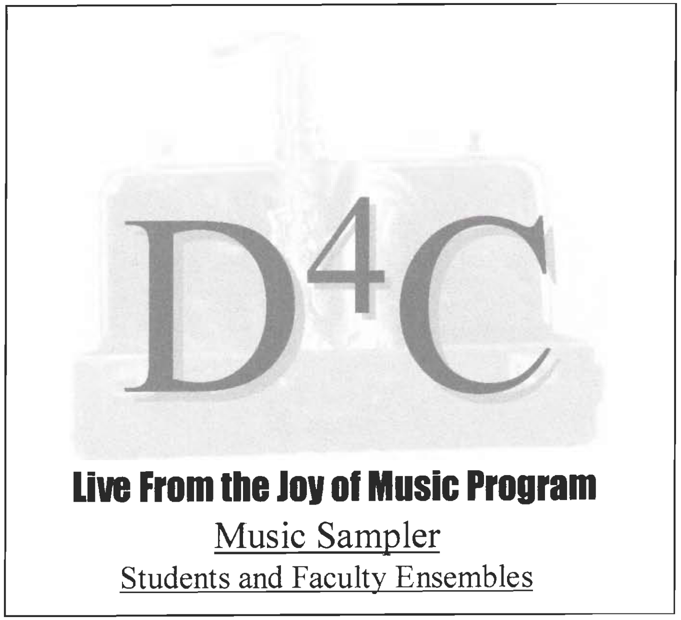
It was necessary for Mr. Blondin to attend this final graphics meeting because he informed Mr. and Mrs. Ardizzone of the photo effects he could produce using the PhotoShop Adobe software program. With the Ardizzones aware of Mr. Blondin's abilities, it was decided that one of the photographs selected by Mr. Blondin be used for an image to be printed onto the sampler disc itself.

Mr. Blondin agreed to set aside some of his time to graphically manipulate and text edit the JOMP photographs. He also printed the final CD templates free of charge. All of these final images were saved onto a zip disk for storage and retrieval ease.

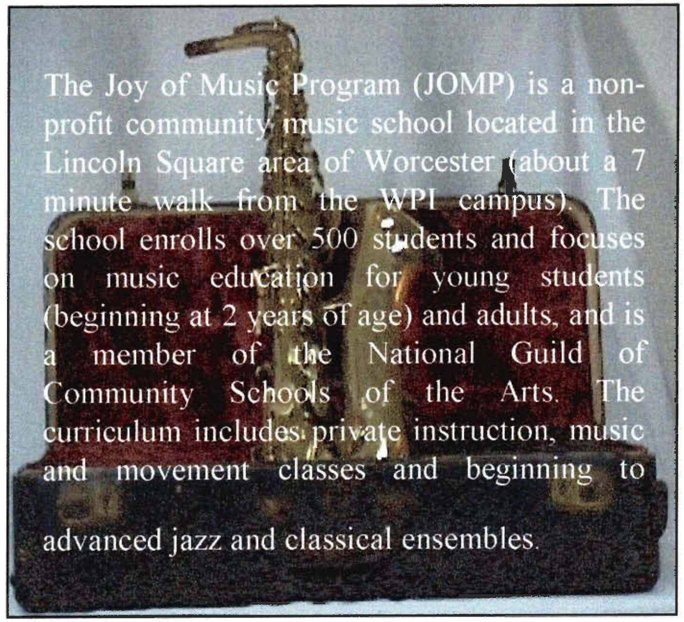
The Ardizzones selected the final CD photos and text found in **Appendix E**

4.3 APPENDIX C: Sampler Template Mockups

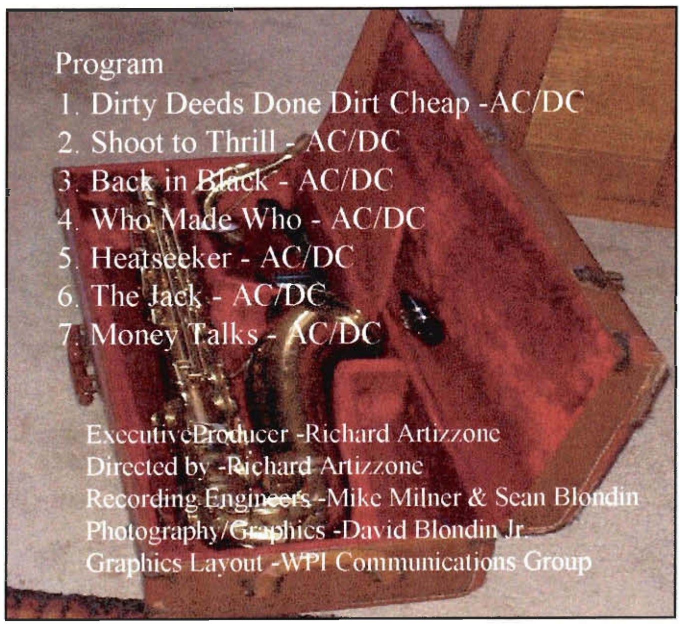
Front Cover



Back Panel



Inside Cover

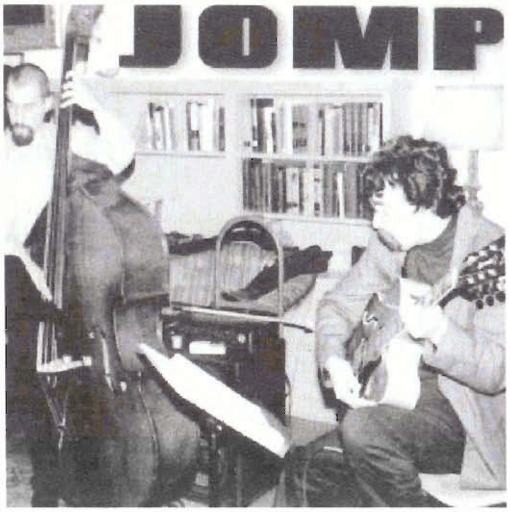


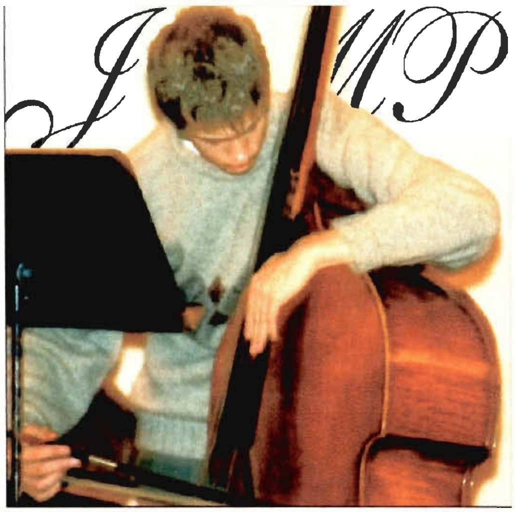
If CD Bed is Clear

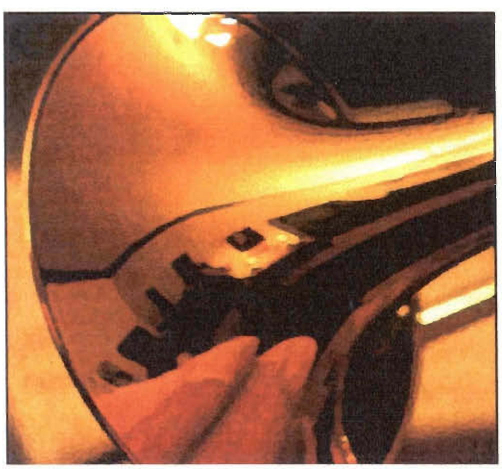
4.4 **<u>APPENDIX D:</u> Examples of David Blondin's Graphic Manipulations**

Edited

Unedited







4.5 **APPENDIX E: Final CD Cover Graphic Templates**

***Front Cover Templates** – all photographs courtesy of David Blondin*

Template 1. Front Cover

Template 3. Inside (next to template 2)

Template 2. Inside (opposite to template 1)

Template 4. Inner Cover (opposite to template 3)

***Back Cover Templates** – all photographs courtesy of David Blondin*

Template 5. Transparent CD Tray Template (found under CD tray)

Template 6. Rear Cover

4.6 **APPENDIX F: Sampler Text Templates**

Each of the following templates was sized, with no borders, to directly overlay the photo templates. The number of each text template corresponds with number of its matching photo template.

The text templates were written by Richard Ardizzone.

Template 1.

Joy of Music Program

A Musical Sampler

Students:

Chamber Music
Chamber Orchestra
Performance Jazz Ensemble

Faculty:

Jazz Sextet
Chamber Music

Template 6.

Chamber Music (student)
Trio in Bb Op. 11 Beethoven

1. **Allegro - Con Brio**

Chamber Music (faculty)
Trio Sonata in C Quantz

2. **Affettuoso**
3. **Larghetto**
4. **Vivace**

Chamber Orchestra (student)
Concerto Grosso Handel

5. **Allegro**
6. **Allegro Moderato**

Chamber Orchestra (student)
Romanian Dances Bartok

7. **Joc cu Bata**
   Braul
   Buciumeana
   Poarga Romanbasca
   Maruntel
   Maruntel

Performance Jazz Ensemble (student)

8. **Midnight Mambo**
   Oscar Hernandez
9. **Samba de los Gatos**
   Mike Steinel

Faculty Jazz Sextet
Music by Jerry Sabatini

10. **Serenity**
11. **Tune Formerly Known As**
12. **You'll Never Know**

Template 2.

Chamber Music (student)
Ages 14 to 16
Tim Terranella, coach
Keithe Baggett, piano
Liana Popkin, flute
Paul Wright, cello

Chamber Orchestra (student)
Ages 11 to 16
Tim Terranella, Conductor
Violins: Matt Guy-Hamilton, Kristen Koch, Emily Mott, Jenna Nordberg, Cecelia Servatius, Christopher Widak, Shauna Zbikowski
Violas: Jacob Appelbaum, Alexandra Pope
Cellos: Amelia Arnold, Alissa Mott, Luke Rosseel, Caroline Reiner-Williams
Double Bass: Max Zeugner
Flutes: Mandy Couture, Avital Mendelson, Stacey Mott, Alison Wade
Clarinet: Eviatar Frankel
Harpsichord: Marina Minkin (faculty)

Chamber Music (faculty)
Jerry Bellows, *alto recorder*
Marina Minkin, *harpsichord*
Tim Terranella, *flute*

Performance Jazz Ensemble (student)
Ages 14 to 17
Rich Ardizzone, *coach*
Corey Bernhard, *piano*
Joshua Filgate, *trombone*
Ben Glaser, *alto sax*
Stephen Schwall, *drums*
Mark Zaleski, *soprano sax*
Max Zeugner, *double bass*

Faculty Jazz Sextet
Jim Allard, *alto sax & flute*
Rich Ardizzone, *trombone*
Mike Connors, *drums*
Rich Falco, *guitar (guest)*
Thomson Kneeland, *bass (guest)*
Jerry Sabatini, *trumpet*

__Template 3.__

The Joy of Music Program, a community music school located in Worcester, Massachusetts, enrolls over 550 students from 2 to 75 years of age. The student and faculty performers on this CD personify the school's philosophy of celebrating music as a joyful part of daily life.

Whether studying music to prepare for a performance career or simply for the joy of it, students are guided by outstanding faculty carefully chosen for their own high level of performance ability as well as their enthusiasm for teaching. Students of all levels are encouraged to develop a healthy relationship with music based on a serious commitment to their own growth and enjoyment as well as to their performance partners.

We hope you enjoy this sampler of student and faculty music from the Joy of Music Program.

Recorded in early December of 1999 at the Joy of Music Program as the students rehearsed for their First Night Worcester performances. Thanks to: Mike Milner and Sean Blondin, WPI interns, for recording and master editing, Dave Blondin for photography, Pat Issacson for graphics and Rich Falco, WPI faculty supervisor for his wise and gentle guidance.

Bibliography

Acoustical Analysis of Spaulding Recital Hall, WPI IQP, Michael Andrews, Michael Corbin, Andrew David, Kyle Warren, 1994.

AKG Acoustics C 100 S +PPC 100 (Polar Pattern Converter) Manual, AKG Acoustics, Harman International, Nashville, TN, 1997.

Bartlett, Bruce and Jennifer, On-Location Recording Techniques, Focal Press, Butterworth-Heinemann, Woburn, MA, 1999.

Beranek, Leo L. Music, Acoustics & Architecture, John Wiley & Sons, Inc., New York, NY, 1962.

Computer Music, Issue 10, U.K. version. September 1999.

Digital Performer- Guide to Digital Audio, Mark of the Unicorn, Inc., Mark of the Unicorn, Inc., Cambridge, MA, 1996.

Electronic Musician, Vol. 15, No. 9, September 1999.

EQ – Professional Project Recording & Sound, Volume10, Issue 8, August 1999, Craig Anderton, "What's the Deal with Digital Mixers?" (44).

Finale Allegro, Volume 2 Encyclopedia, 1.0 for Macintosh, Coda Music Technologies, Eden Prairie, MN, 1993.

Fry, Duncan Live Sound Mixing, Roztralia Productions, Second Edition, New York, NY, 1992.

Home Recording, Vol. 2, Num. 4, October 1999, Massey, Levine, Gelfand, Prown, Philbrick, "Format Wars!", (42)

Huber, David. Runstein, Robert, Modern Recording Techniques, Sams Publishing, Carmel, IN, 1989.

Lathi, Bhagawandas Pannalal, Signal Processing & Linear Systems, Berkeley-Cambridge Press, Carmichael, CA, 1998.

Mix, Vol. 23, No.9, September 1999; Alberts, Randy "Living Large: New Large-Diaphragm Condenser Microphones".

Mix, Vol. 23, No.9, September 1999; "New Software/Hardware for Audio Production"

Olson, Harold Modern Sound Reproduction, Van Nostrand Reinhold Company, First Edition, Huntington, NY, 1972.

ProTools: System Installation Guide, Digidesign, Avid, Palo Alto, CA, 1997

Recording: The Magazine for the Recording Musician, September 1999, Case, Alex. "Nuts & Bolts: Multitrack Recorders" (46).

Recording: The Magazine for the Recording Musician, September 1999; Ross, Bob. "Horns of Plenty: Sections" (56)

Traylor, Joseph Physics of Stereo/Quad Sound, Iowa State University Press, First Edition, IW, 1977.

URL References

http://www.midi.com/

http://www.JeepJazz.com/handbk.html/

http://www.digidesign.com/

http://www.digido.com/depthessay.html/

http://recording.hostway.com/micing/

http://homepages.packet.net/jgm/mic-faq.txt

http://www.digital-recordings.com/publ/pubrec.html

http://eklektix.com/dat-head/digests/

http://www.outersound.com/osu/recording/

http://tape.com/techinfo/livetape.html

http://www.american-digital.com

http://www.computermusic.co.uk

http://www.emusican.com

http://www.mixonline.com

http://www.homerecording.com

http://www.mackie.com

http://www.akgacoustics.com