Date: July 7, 2021

DMA Option (circle): 2 [thesis] or 3 [scholarly essay]

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THE ELECTRIC SAXOPHONE:
AN EXAMINATION OF AND GUIDE TO ELECTROACOUSTIC TECHNOLOGY IN
CLASSICAL SAXOPHONE REPERTOIRE

BY

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SCHOLARLY ESSAY

Submitted in partial fulfillment of the requirements
for the degree of Doctor of Musical Arts in Music
with a concentration in Performance and Literature
in the Graduate College of the
University of Illinois Urbana-Champaign, 2021

Urbana, Illinois

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ABSTRACT

Throughout history, advances in technology have impacted classical saxophone repertoire. As new electroacoustic tools for composition were created, classical saxophone repertoire showcasing these new creative technologies also emerged. Many of these works omit technical details, however, and instead rely on a performer’s technological knowledge to fill in the gaps. Because of this, performers without the prerequisite technological experience can feel intimidated and shy away from electroacoustic saxophone repertoire. Performance of this music plays an important part in its preservation, and this project is designed to encourage performance of electroacoustic saxophone works. The opening portion of this essay outlines important developments in electroacoustic technology, key electroacoustic works, and how those technologies intersected with saxophone repertoire. This is followed by a thorough guide that breaks down how to approach electroacoustic works and lowers the barrier of entry to electroacoustic saxophone performance.
ACKNOWLEDGEMENTS

I would like to start by thanking the University of Illinois Urbana-Champaign School of Music and the Graduate College for their academic and financial support of my time at UIUC. Additionally, I am thankful to the Bowling Green State University Music Library, the Bill Schurk Sound Archives, and Ritter Library at Baldwin Wallace for hosting me and providing every material I used for my research in an extraordinarily timely, organized, and welcoming way. In particular, thanks to Patty Falk (BGSU) and Laura D’Amato (BW) for coordinating directly with me to help complete the dozens and dozens of requests for materials. Similarly, thank you to the members of Baldwin Wallace faculty and staff that allowed me to use their facilities and resources to complete this degree, including Craig Reynolds, Bill Hartzell, Charles Young, and Susan Van Vorst. I also express the sincerest appreciation for Jenny Philips (UIUC) and her phenomenal ability to quickly problem solve – she is truly an amazing administrator that solved many of my seemingly insurmountable challenges and mishaps during the DMA processes. I am so grateful to my committee, Debra Richtmeyer, Eli Fieldsteel, Chip McNeill, and Megan Eagen-Jones, for their insight and guidance throughout the entire DMA process. A special thank you to Debra Richtmeyer for her constant inspiration and wisdom.

Many friends in my personal life were also extraordinarily helpful throughout the process. Thank you to my many Cleveland friends who have supported me and were some of the most welcoming people when I first moved to Cleveland. Jill Klatt and Eric Væbn have been unbelievably helpful in getting me to relax and keeping me well fed. A special thank you to my remote friends from New York and Georgia, Hannah Pearson and Kathryn Theodosakis, for help with preparing for the exams and proofreading my writing. And I express my deepest gratitude to
Lauren Camacci, Ph.D. for her extraordinary intellect, selfless eagerness to help, and profound impact she had on me personally and academically.

Finally, I would like to thank my family. They have seen me throughout this entire process and have been there for me every step of the way. Without hesitation, they have sacrificed so I can thrive, and I am forever grateful.
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CHAPTER 1: Historical Context

Throughout the course of musical history, technological developments and advancements in musical repertoire have been intrinsically linked. For example, with the development of the pianoforte, new technical and dynamic possibilities arose, alongside repertoire to highlight these new features. The development of the clavichord lent itself to the Empfindsamer Stil, inspiring repertoire that highlighted not only the cultural sentiments of the time but also the instrument’s expressive capabilities.¹ In the 20th and 21st centuries, developments in electronic music and computer technology have influenced the canon of saxophone repertoire. Today, electroacoustic saxophone music is an integral part of the saxophone community as evidenced by the frequent electroacoustic performances found at saxophone conferences and competitions around the world.

1.1 The Advent of Electroacoustic Saxophone Compositions

Advancements in musical instrument technology in the 20th and 21st centuries have been primarily electronic, with compositions and techniques evolving alongside the composer’s ability to control electronics.² The first evidence of electronics influencing saxophone repertoire can be found in work done by composer Terry Riley in the early 1960s. Although Riley’s first work for saxophone is debated, Jean-Marie Londeix’s Guide to the Saxophone Repertoire (one of the most comprehensive sources on saxophone repertoire) lists Riley’s composition Poppy NoGood and the Phantom Band (1963) as the first electroacoustic saxophone work.³ In a program for an event

at Steinway Hall in New York City on April 25th and 26th, 1968, Riley writes about this work, originally titled “Solo Time Lag Music for Soprano Saxophone”:

All the material that I am playing subsequently recycles and combines in an accumulative manner. In this way many generations of the material can be quickly built up without having to add each track one at a time, therefore adapting itself naturally to use in live performance. This is the free list of all my recent work as the automatic ordering of the material in the time lag accumulation process allows me to play quite complicated material which is then arranged into loops and recycled… I have written no scores for this music as so far it has all been governed by an intuitive relationship developed between me and the machines. I do have a catalogue of material which I use as a basis for these improvisations and am constantly adding new patterns. However, I want to keep the music in the tradition of unwritten improvised music.⁴

The development of his self-described “time-lag accumulation” technique, and the unwritten “improvisational tradition” Riley identifies as an influence, were first heard in his earlier work, Music for The Gift. Riley wrote Music for The Gift for an experimental play during his time in Paris in 1963. The work involved Riley recording members of the Chet Baker Quartet, also in Paris at the time, performing a rendition of So What by Miles Davis. He used two tape recorders connected by a single tape loop. One recorded, while the other reproduced the signal, allowing him “to control the time between the initial playback and echo.”⁵ This first collaboration with Chet Baker and his experimentation with the time-lag accumulation technique would lay the groundwork for Poppy NoGood and the Phantom Band. Riley’s original recordings of Poppy NoGood and the Phantom Band are of himself playing the soprano saxophone into the time-lag accumulator, stating that he “didn’t want to tell the saxophone

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⁴ Terry Riley “Poppy Riley Poppy NoGood’s Phantom Band,” program for Solo Time Lag Music For Soprano Sax (New York: Steinway Hall, April 25, 1968).
player to do this or that… I didn’t see any choice but to see if I could learn the saxophone well enough to do it.”

Other early works for saxophone and electronics incorporated similar delay effects and cumulative tape delay systems, such as Dorian Reeds (1964) by Terry Riley and Saxony (1978) by David Tenney. Saxony, which was commissioned and premiered in 1978, uses a similar accumulating effect, but unlike Riley, harmonically organizes the sound into a “stochastic canon,” allowing for some improvisation, but carefully controlling tonality, adding a new dimension to Riley’s existing technique. Tenney achieves his “stochastic canon” effect by first accumulating the sound of the lowest Eb on baritone saxophone, then increasing the intensity of improvisation as the performer moves up the overtone sequence. The highest overtones in the series are to be played on alto and soprano and indications for just intonation of the pitches is included in the score.

“There are no theoretical limitations to the performance of the computer as a source of musical sounds, in contrast to the performance of ordinary instruments.” –Max Mathews

1.2 The Digital Revolution

During the late 1950s and early 1960s, musical programming languages were being written to facilitate the use of computers as musical instruments. One of the pioneers of these early programming languages was Max Mathews, an engineer at Bell Labs. In 1957, Mathews

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7 James Tenney, Saxony: for Saxophone Player(s) and Tape-Delay System (Baltimore: Smith Publications, 1986), 1.
was able to use the programming language he had created to run the first program to generate sounds directly from a computer. The language he created was named MUSIC, and it would develop into an influential series of languages and programs collectively known as MUSIC-N.

These programs, and Mathews’s work with them, strongly influenced the work of Barry Vercoe, another pioneer in computer music. Vercoe expanded on Mathews’s work, which led to the development of MUSIC 360 in 1969, then MUSIC 11 in 1973. One noteworthy aspect of Music 11 was that it was coded primarily in C, a new programming language at the time. Like the MUSIC languages, C was also developed at Bell Labs. C and its derivatives (C+, C++, C#, etc.) are still some of the most popular programming languages today. In 1986, Music 11 was ported to C and named Csound, which is still widely used today in electroacoustic performance and composition.

The advent of the personal computer revolution in the 1980s introduced an entirely new level of accessibility, portability, and creativity with computer technology. Computer technology was flourishing, and the sense that digital electronics might dominate the reproduction and synthesis of music was taking hold. The technological developments during this era directly intersected with saxophone repertoire to create new types of electroacoustic music intended for saxophone and computer. Three notable advances during this era include the introduction of music-specific programming languages enabling real-time performance and interaction with a live performer; the development of MIDI, enabling simpler connectivity and increased

compatibility between different types of hardware and software; and advances in electronic wind instrument technology.

1.3 Music-specific Programming Languages

MAX was one of the most important music-specific programming languages of the 1980s and was created by Miller Puckette, a student of Vercoe, and named after Max Mathews.\(^{13}\) The first version of MAX, written in the C programming language, as developed in 1987 ran on the Macintosh computer.\(^{14}\) The main advantages of MAX over other programming languages at the time were its graphical user interface (GUI) and ability to processes parallel tasks, such as triggering external synthesizers. The GUI simplified the writing of MAX programs (called “patches”), using an interconnected network of boxes and connecting lines to represent each virtual object and its signal connections. Additionally, MAX allowed real-time inputs to trigger commands without interruption of concurrent tasks, enabling a new level of interactivity between performer and computer.\(^{15}\)

At that time, MAX was only able to control external synthesizers, and the ability to create audio signals from scratch using electrical signals, also known as sound synthesis, had not yet been integrated into MAX. Puckette’s next goal was to add sound synthesis functionality similar to that of programs such as Csound, a program with extensive sound synthesis capabilities.\(^{16}\) In 1997, an extension for MAX known as MAX Signal Processing (MSP) was released, adding integrated sound synthesis capability and additional live audio processing resources.\(^{17}\) This extension has become an inseparable part of MAX, which today is commonly referred to as

\(^{13}\) MAX is also commonly written as Max. To avoid confusion with Max Mathews, the program MAX will be capitalized in this essay.


MAX/MSP.\textsuperscript{18} MAX/MSP and Csound, which are still in use today, are merely two examples of a large family of audio programming software platforms that have come into existence in the past half-century.\textsuperscript{19}

1.4 MIDI

The Musical Instrument Digital Interface (MIDI) was launched in 1983 and remains a critical part of electroacoustic performance and composition. MIDI is a standardized protocol for digital communication of musical information.\textsuperscript{20} These signals are commonly used to transmit information about musical pitch, dynamics, duration, sustain pedal, vibrato, and clock signals (for tempo), among other signals.\textsuperscript{21} MIDI signals are commonly generated by physical MIDI control devices, which exist as keyboards, drum sets, wind controllers, touch pads, foot pedals, and many others. The actual timbre of sounds generated by MIDI data, however, are independent of the controller and the MIDI data. Instead, the receiving device determines the quality of the sound. MIDI provided standardization that allowed compatibility between different instruments and computers, leading to new horizons of music performance.

1.5 Advances in Electronic Wind Instrument Technology

Although early electronic wind instruments such as the Lyricon and Varitone existed before 1987, they were purely analog devices.\textsuperscript{22} Alongside MIDI and music-specific programming languages, advances in electronic instrument technology led to the development of electronic wind instruments (EWIs) that were able to send MIDI signals. With these new


\textsuperscript{19} Other common programs include Pure Data (also developed by Miller Puckette) and SuperCollider.


\textsuperscript{22} The Varitone was an acoustic instrument with an attached signal processor that either added and amplified analog effects or could be used completely acoustically.
instruments, wind players were able to transmit MIDI data and produce sound by performing on an instrument that closely resembled a saxophone, clarinet, flute, or valved brass instrument, rather than a MIDI keyboard. Musical dynamics and note durations were controllable via breath, pitch could be adjusted with a bite sensor, and fingerings mirrored that of these instruments’ acoustic counterparts. The earliest MIDI controllers were the Yamaha WX7 and the AKAI EWI1000, released in 1987.

1.6 Intersections with Saxophone Repertoire

These advancements greatly facilitated creative possibilities, leading to saxophone works that would not have previously been possible. One of the first pieces for saxophone to incorporate these new technologies was Morton Subotnick’s In Two Worlds. Composed in 1987, this work was originally conceived for solo alto saxophone, wind controller, computer, and orchestra, and it is the first classical composition written for an electronic wind instrument, specifically for the Yamaha WX7 Wind Controller. Subotnick was quoted to say of his work:

“…the Yamaha WX7 Electronic Wind Controller is a new musical instrument which allows the performer’s musical gestures to be transformed into digital signals, which can then be translated into specially created sounds. The digital signals can also be read by the computer so that it will be “aware” of the exact location of the performer in the score.”

Between 1987 and today, the decreasing cost of computing power and increasing cultural interest in technology have exponentially expanded the number of works for saxophone and electronics. Additionally, self-publication and the widespread availability of technology have made it nearly impossible to keep an updated comprehensive catalog of all saxophone works that

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include electronics. Those that come close are the Electroacoustic Repertoire Database created by the Society for Electro-Acoustic Music (SEAMUS), the Computer Music Library (COMPEL) created by Virginia Polytechnic Institute and State University, and the Works for Saxophone Resource created by Dr. Sarah Hetrick.27 This essay contributes to existing research into electroacoustic saxophone repertoire by lowering the expertise threshold to this genre. By providing a guide to the specific hardware, software, and best practices of electroacoustic music, this project encourages performers now, and in the future, to take up this genre with confidence.

CHAPTER 2: Guide

Deciphering and understanding the necessary equipment to perform an electroacoustic saxophone piece can be difficult. Many details, like the quantity and type of cables, microphone type and placement, and compatibility of hardware and software, are often omitted from the instructions on how to perform the work. By such omissions, the composer implicitly relies on the performer’s knowledge of, and experience with, the technology required to perform the piece as intended. This may create a situation in which performers with less electroacoustic experience are intimidated or discouraged by seemingly complicated and vague instructions. They may thus, and too often do, abandon or dismiss this genre of music.

One of the primary challenges of approaching electroacoustic repertoire is learning how to deconstruct an electroacoustic piece into its constituent parts and identify required items. This guide is designed to meet that challenge by familiarizing a performer with common equipment, setups, and other considerations for performing electroacoustic saxophone repertoire. In the following pages, guide users should read the introductory material (Signal Path, Amplification and Pre-Amplification Stages, and Using the Flow Chart), and then use the flow chart in the following section to answer a series of questions. Each set of questions and answers in the flow chart has a corresponding section with additional information. The appendices also include information and images to help clarify the referenced equipment.

2.1 Signal Path

An electroacoustic saxophone work often includes a summary, diagram, or technical drawing indicating how to set up a performance of the work. This information is usually found in the first few pages of the score and may include a drawing known as a signal path (Figure 1). A signal path (often used interchangeably with signal flow or signal diagram) is a drawing that
shows the route a signal takes, sometimes passing through multiple devices, including the required routing steps or processing steps in between. In the case of electroacoustic repertoire, this signal is most often an audio signal.\textsuperscript{28} The source of a signal can be created in a number of ways, such as a microphone capturing the sound of a saxophone, a CD player, a tone generator, or a computer. Though an understanding of how the signal moves through different pieces of equipment is critical, most signal paths exclude some of the finer details of how to configure the equipment and how exactly to get the signal to its various destinations. To organize the process, the signal path will be divided into the amplification stage and pre-amplification stage. In Figure 1, we can trace the signal flow from the microphone source to the input of the preamplifier, the output of which is then plugged into the input of the EQ. From there, all the outputs (right side) of processing equipment (EQ, pitch shift, delay, loop machine, and reverb unit) are plugged into the inputs (left side) of the next piece of hardware in the signal flow diagram.

\begin{figure}[h]
\centering
\includegraphics[width=0.5\textwidth]{signal_path_diagram.png}
\caption{Simple signal path diagram for Cinque Nudi for Saxophone and Stomp Boxes by Marco Momi.}
\end{figure}

\textsuperscript{28} Other non-audio signals in electroacoustic music are used to communicate data that can be used operate equipment or trigger events, such as MIDI data.
2.2 Amplification and Pre-Amplification Stages

The amplification stage is the one of the last steps in the signal path, in which the signal is amplified to be played through loudspeakers. Because all electroacoustic pieces require amplification, this stage will likely be similar and interchangeable regardless of the electroacoustic work being performed.\(^\text{29}\) Examples of equipment used in the amplification stage are PA (public address) systems, concert hall and auditorium sound systems, portable stereos, and Bluetooth speakers, among many others. These devices all provide the same function: to reproduce an audio signal at a louder volume.

Often, the type of amplification is determined by factors including room size, equipment availability, and specifications in the score. It is uncommon for saxophone performers to have a personal amplification system that is sufficient; borrowing or renting the correct equipment is usually the best solution when performing these pieces in a venue without a suitable amplification system. Many concert halls and performing spaces, however, have a portable or permanently installed amplification system readily available. Most amplification systems will offer a wide variety of signal connections, but it is important to match the type of input on the amplification system with the correct type of output from the pre-amplification stage.\(^\text{30}\)

There are a variety of audio signal levels that also need to be considered when making connections. Simply put, different types of equipment are designed to send and/or receive different levels of signals. The four primary signal levels are mic level, instrument level, line level, and speaker level. Mic level signals are generated by microphones and require additional

\(^{29}\) This applies primarily to works that only need mono or stereo amplification. Some electroacoustic pieces require 4, 6, 8, or more loudspeakers carefully arranged around an audience and require more complicated setups and additional equipment, but the same principle applies.

\(^{30}\) See Figure 7 and 11 for visual examples of labeled input and output jacks.
amplification through a preamplifier.\textsuperscript{31} If a device input is labeled “Mic” or “Mic In,” it has an integrated preamplifier and is designed to accept mic level inputs. Instrument level signals are primarily generated by electric guitars and should not be plugged into input jacks designed for mic level or line level inputs as they may cause distortion or be exceedingly quiet. Line level signals are signals that are stronger than mic and instrument levels and generally come from powered devices, such as computers, CD players, outputs on mixers, or signal processors. Speaker level refers to high-voltage signals found after the amplification stage, which are transmitted from the amplifier to the loudspeakers. To ensure proper functionality and safety of all equipment, it is important to always be aware of the signal levels and match the signal level of inputs and outputs.

Audio signals can also be transmitted in two ways: via balanced cables or unbalanced cables. Unbalanced cables are most often used to connect guitars and effects pedal and are not suitable for transmission over long distances due to their susceptibility to electrical interference. Balanced cables are less susceptible to electrical interference, suitable for longer distances, and more common in electroacoustic applications. Most output jacks are labeled balanced or unbalanced or can be determined by looking in the device’s manual.

For the purposes of this paper, the pre-amplification stage refers to the part of the signal flow in which signals originate and are processed before they enter the amplification stage. The pre-amplification signal path varies widely between pieces, and hardware needed to connect everything depends on the type of signal processing required, the presence of fixed-media playback, what (if any) inputs are required from the performer during a performance, and the types of hardware needed to connect everything. The flow chart shown in Figure 2 was

\textsuperscript{31} Not to be confused with the subsequent section which refers to all the signal processing and routing before the amplification stage as the “pre-amplification stage.”
developed to present relevant information in an organized manner, and to assist with separating a work into its constituent parts.

2.3 Using the Flow Chart

The flow chart (Figure 2) presents a series of binary questions meant to help the performer quickly assess and identify the needs of an electroacoustic work. Information relevant to answering the questions can be found in the related sections. This guide is intended to be approached not as a book or novel, but as a sort of user manual, with the flow chart serving as a map.
2.4 The Flow Chart

Figure 2. A flow chart intended to help visualize the requirements of an electroacoustic work and organize such a work into its constituent parts.
2.5 Fixed Media

The first step is determining whether the piece contains any *fixed media*, which refers to any type of pre-recorded material meant to be played back as-is. Fixed media can take two forms: segmented and non-segmented. Segmented fixed media exists in multiple pieces or multiple files, and often requires input from a performer or assistant to advance through the audio fragments (input methods will be discussed in a later section). Works that use non-segmented fixed media contain a single file or track that is played, uninterrupted, alongside the live performer; in this case, no additional input is required once playback has begun.

Key indicators are helpful in determining whether the piece contains fixed media, and whether it is segmented. If the title or subtitle is “for saxophone and tape,” or if a CD with audio tracks is included, it can be presumed that the work is for non-segmented fixed media. If the work is listed as “with electronics,” it may explicitly say near the beginning of the work if it is for a single prerecorded track (examples include *Music for Alto Saxophone & Electronics* by Pablo E. Furman (1995)\(^\text{32}\) and *Billie* by Jacob TV\(^\text{33}\)).

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If the work is for segmented fixed media, the fixed media part will usually be notated in the score with cues to advance through the segments. An example of segmented fixed media with cues can be seen in *Dissidence 1b* by Christophe Havel (Figure 3).

Figure 3. *Dissidence 1b* by Christophe Havel showing notated fixed media cues as boxed numbers.
2.6 Specialized Hardware and Software

For the purposes of this essay, specialized hardware refers to any electronic equipment beyond a standalone music player (such as a CD or MP3 player) or a computer. Specialized software refers to any computer program beyond a basic media player (such as VLC, Windows Media Player, QuickTime Player, etc.). If the piece does not require specialized hardware or software, it will most likely contain non-segmented fixed media and will not require amplification of the saxophone. The system for playback of the track (CD player, MP3 player, computer playback) should be connected to the amplification stage directly.

Using a cable with two male RCA connectors on one end (Figure 4) and a single 1/8-inch male TRS connector (Figure 5) on the other is a common way to connect playback devices to a PA system, as most PA systems and mixers have RCA jacks, and most non-specialized playback devices use 1/8-inch TRS jacks. Additionally, some lecterns in

Figure 4. RCA Connector(s). Used for carrying a one-channel signal on each connector, however they are most commonly found in a pair (as shown in the image). RCA jacks are commonly found on mixers, labeled “tape in/out” or “2tk in/out.” Usually terminated in another pair of RCA connectors or a single 1/8-inch TRS connector. Also known as phono connector.
concert halls, classrooms, and performance spaces are already connected to the amplifiers for the room or hall and are already outfitted with a male 1/8-inch TRS cable that can be used to directly plug into the playback device.

A work is for specialized hardware if it calls for any hardware such as microphones, mixers, effects pedals, digital signal processors (DSPs), delay machines, tone generators, or foot pedals, but not specialized software. “Specialized hardware only” electroacoustic works can take on any combination of three primary forms:

1) simple amplification of the saxophone signal
2) live hardware processing of the saxophone signal
3) tone/noise generation

For simple amplification of the saxophone signal alongside a prerecorded track, the microphone capturing the saxophone sound and the playback device should be connected to a mixer (Figure 7). The mixer can then be used to adjust the level of the saxophone and audio track separately to ensure the proper balance between them. Most professional microphones use an XLR cable (Figure 6) to connect to other hardware, such as mixers. It might also be possible to connect both the microphone and playback device Figure 5. 1/8-inch Connector (TRS). Used for carrying a stereo signal. They are commonly used for MP3 players, phones, and car stereos. Also known as a mini plug, headphone jack, or 3.5mm jack. Notice the the three contact points, creating a tip (T), a ring (R), and a sleeve (S).

Figure 6. XLR Cable. Used for carrying a balanced, one-channel audio signal. Commonly used for connecting microphones to other devices, and also used for connections between monitors, audio interfaces, and PA systems. Most commonly have a female end (left) and a male end (right).
directly to the amplifier if a mixer is not available, as some amplifiers, most often PA amplifiers, have mixing capability and accept both mic and line level inputs.

Figure 7 shows an example of a typical mixer. Letter A (red) encloses the mixer inputs. Letter B (green) encloses the mixer outputs. The main outputs (mix output) would ordinarily be connected to the main amplifier, while CR output (control room) and phones could be used as a signal path for performer monitoring. Letter C outlines an output (aux send) used to send an audio signal to an effects pedal or digital signal processor. The output signal of the effect should return through the jack labeled aux ret. The dotted box outlines the inputs, equalization, balance, and level of Channel 1. Channels are usually aligned vertically, and each input should have its own channel.
Figure 7. Diagram of inputs and outputs on a typical mixer.
If a piece requires specialized software, it will most likely need specialized hardware as well – usually an audio interface (Figure 11). The purpose of the interface is to take the audio signal from a microphone or other analog source and translate it into a signal that the computer can understand (i.e., a digital signal). The computer can then manipulate the sound, synthesize new sounds, or add fixed media to the original signal and output that signal to the audio interface, which would continue to the amplification stage (or mixer). Common software programs used for electroacoustic performances include MAX/MSP, Pure Data (Pd), SuperCollider, Csound, and Kyma. Patches that run on these platforms typically generate sound, manipulate sound, or play pre-recorded audio files. The score most often specifies the needed software. A microphone can connect directly to the XLR input of an audio interface, and the interface is usually connected to a computer via USB cable. Most audio interfaces have 1/4-inch TRS outputs (Figure 8), to be connected to the amplification stage.

Figure 11 illustrates a typical audio interface. Letter A shows a combination port that can be used to connect a XLR cable or a 1/4-inch TS connector (Figure 9). As indicated, it is compatible with microphone level inputs, or instrument level inputs. Letter B shows the phantom power switch (labeled 48V). Some microphones require extra power which can

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34 The capabilities of these software extend well beyond the functions listed here and can also be used to incorporate other elements such as video and interactive components.

35 See Appendix A for visual examples of the different types of USB cables.
be provided by enabling the phantom power
switch. Some microphones can be damaged or
destroyed if phantom power is enabled, and
should only be provided if required. Letter C
shows the USB B jack used to power and
transmit data to a computer. Letter D shows the
input and output jacks. It is important to be
mindful of the directional flow of MIDI data when making such connections (Figure 10), and to
ensure that the output of some MIDI device is connected to the input jack of the interface, and
vice-versa. Letter E shows two balanced outputs used to connect to a mixer or an amplifier.
Letter F shows a stereo headphone jack that can be used to monitor signals from the computer or
signals from the inputs on the front of the device.

Figure 9. 1/4-inch Connector (TS). Used
for carrying an unbalanced, one-channel
audio signal. Cables with these ends are
commonly used with guitars, guitar
pedals. Notice how the connector only
has two discrete contact points, creating a
tip (T) and a sleeve (S).

Figure 10. MIDI Cable. Used to carry MIDI signals between MIDI capable devices.
Identifiable by its 5 pins and notched enclosure. Although uncommon, they are still used to
control a variety of older, MIDI only hardware. Both ends are identical.
Figure 11. Front (above) and rear panel (below) of a typical audio interface.
2.7 Real-Time Audio Processing

Specialized hardware and software are commonly used to perform real-time audio processing. Real-time audio processing involves taking the signal from the saxophone and manipulating it instantaneously or with very little latency.  

Hardware that is specially designed for this task most commonly takes the form of effects pedals or digital signal processors. Common methods for manipulation of a signal including adding reverb, delay, distortion, looping functions, or pitch shifting. Some works have a single static effect present throughout (Figure 12 and 13), while others contain a multitude of changing effects (Figure 14).

For live signal processing via hardware only, the microphone capturing the saxophone sound will need to be connected to the effects pedal or DSP. If the effect pedal or DSP has an XLR input jack indicated as “mic in,” it is safe to connect the microphone directly to it. Most effects pedals are designed for use with guitars, however, and often only have a 1/4-inch TS input and a 1/4-inch TS output. One of the most common and reliable solutions uses the following simplified signal path: mic → mixer → effects pedal → mixer → amplifier.

This signal path is simple but requires that the mixer have auxiliary send and auxiliary return paths (abbreviated as aux sends or FX Send/Return). The microphone will be connected

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36 Latency in this context is referring to the amount of time required for a signal to pass through an entire signal chain. See latency in “Other Tips and Additional Considerations.”

37 See Figure 7 for a visual example.
directly to a mixer input, using an XLR cable. Then, a 1/4-inch TS cable can be connected from the AUX Send of the mixer to the effects pedal or DSP. A 1/4-inch TS cable will then be used to connect the output of the effects pedal or DSP to the AUX input of the mixer. Balance between the unprocessed saxophone signal, the processed saxophone signal, and any other signal (e.g., from a prerecorded track) can then be adjusted on the mixer and sent to the amplifier. *Cinque Nudi for Saxophone and Stomp Boxes* by Marco Momii and *Grand Larceny* by Joseph Waters (Figure 12) are examples of this setup.

![Figure 12](image_url)

Figure 12. Page three of *Grand Larceny* by Joseph Waters showing the saxophone staff on the score and how the sound should be processed.

Manipulating a signal in real-time through software is also common. In fact, many works that require older, obsolete, or difficult-to-find hardware or software programs use programs such as MAX/MSP and Pure Data to emulate or recreate these technologies. In fact, nearly all works that require specific hardware processing can be emulated in software. For instance, the now obsolete time-lag accumulation technology required for pieces such as *Saxony* by James Tenney, can be easily digitally replicated using MAX/MSP. Many of these programs have vast communities and tutorials outlining how to build certain virtual devices (e.g., delay devices, distortion, and looping effects) in their programming environment. Although not an issue for most works, routing audio signals through software introduces latency. Yet, works that require extremely precise alignment, such as in Figure 13, can be greatly affected by high latency.
PREPARATION:

Tuboing * tuned A3 (plastic tube: longness 76,7 cm / diameter 4,5 cm) to be used in the last movement.

ELECTRONIC DEVICES:

- loop machine
- pitch shift unit
- reverb unit
- delay unit
- cables to connect MIDI controller and all MIDI devices (such as the pedal)
- one speaker
- microphone

For any request about materials, please contact the composer or the publisher.

This piece has been commissioned by Radio France for the series "Alla breve"

Figure 14. Excerpt from Cinque Nudi for Saxophone and Stomp Boxes by Marco Momi showing the list of electronic devices required for performance.
2.8 Interactivity

If the hardware or software will require physical inputs from a performer during a performance, such as pressing spacebar on a computer, playing specific trigger notes, or depressing a foot pedal, the work is considered interactive. If the electronics are set at the beginning of the work and do not require real-time interaction during the performance, then they are considered “set and forget.” Most works that require real-time actions from the performer have indications in the score to show when an input from the performer is needed. Works that use live processing but do not require physical input often have either the composite sound notated or what the overall effect should be, such as with Anthony Fiumara’s Redshift (Figure 13).

With “set and forget” electronics, the effects and/or prerecorded track are not changed and continue uninterrupted throughout the entire piece. An example can be found in Figure 13, in which the saxophone signal is processed with a constant amount of delay that does not vary throughout the piece. If the work is interactive, however, and requires physical inputs during the
performance, effect pedals are one of the most common methods of interaction. Effects pedals are designed to allow the signal to pass through, unprocessed, when the pedal is disabled. An example of a work that does not require specialized software but is interactive is *Naica* by Viet Cuong (Figure 15), which uses a delay pedal that the performer periodically turns on and off throughout the work.

![Figure 15. Page one of *Naica* by Viet Cuong showing the required inputs to the delay pedal.](image)

Interactive works take on two primary forms:

1) the performer directly interacts with the electronics
2) another performer activates the electronics

When communicating with a computer, a foot pedal allows the performer to control events (e.g., triggering a sound, advancing through fixed media, or starting/resetting a timer). Unlike effect pedals, foot pedals only send out a digital signal to another device instead of performing live processing onboard. There are a wide variety of foot pedal solutions for communication to a computer, each with their own advantages and disadvantages. Some common foot pedal solutions:

1) **Standalone MIDI foot pedals** offer a wide variety of MIDI functionality. If planning on playing or even composing interactive works, this solution offers more control options than are likely needed (e.g., expression pedals). Multi-pedal boards tend to be
more expensive than other options, and some (such as the Behringer FCB1010) require that the audio interface have MIDI inputs and outputs. Others, such as the Keith McMillen SoftStep, connect via USB.

2) **Single button USB pedals** are a low-cost option that can be configured to work with programs like MAX and SuperCollider. Most USB foot pedals include software that can be used to bind the foot pedal to different keystrokes (e.g., the spacebar is a common key for advancing through electroacoustic pieces). Compatibility can be an issue, however, if the manufacturer does not include key binding software or if the foot pedal is not configurable.

3) **DIY foot pedals** are a create-your-own option best for those on a very tight budget but who have the requisite time and tools. Many guides exist to help to guide DIY builders, including Cycling74 (publishers of MAX/MSP). It is often a functional solution, though it may not look as professional onstage and might not be as reliable as commercially available pedals.\(^{38}\)

4) **Keyboard sustain pedals** are one of the most popular and low-cost solutions. Sustain pedals are easy to find and work with almost any MIDI keyboard that has a 1/4 sustain pedal input jack. When the sustain pedal is connected to the keyboard and the keyboard then connected to the audio interface (or directly to a computer), a MIDI signal will be sent to the computer when the sustain pedal is pressed. The software can then look for the sustain pedal signal to trigger events in the program. If the keyboard does not have USB capabilities, then the audio interface must have MIDI capabilities.

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inputs/outputs and the MIDI keyboard must be connected to the interface, even though the keyboard itself will not be used.

5) **Bluetooth pedals** are commonly used to advance pages when performing from a digital device. They can also be used to send signals to a computer. Most, such as the Airturn BT200, have an option to send a spacebar, enter, or another keystroke signal when a pedal is depressed. This pedal option requires that the computer have Bluetooth capability. Relatively inexpensive USB Bluetooth adapters are commercially available as well.

Another type of interactive work responds to sound inputs instead of physical inputs. Known as score-followers, these programs synchronize live electronics with acoustic instruments and are usually programmed directly into the software patches by the composer. These programs use acoustical analysis to “listen” to a performer and trigger events accordingly. Although intended to reduce the burden on a performer during a performance, these programs can be inaccurate or awkward and are susceptible to errors due to changes in acoustic environment, tuning, and volume.

2.9 General Tips and Considerations

**When rehearsing electroacoustic music**, have someone sit in the audience and listen for balance. Like any other performance, it is desirable to achieve a balance between the two parts, and for the electronics to remain at comfortable listening level. It is also important that the performer be able to clearly hear the electroacoustic part during the performance through headphones or an onstage monitor. After levels have been set during rehearsal, a photograph of

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all the settings and stage setups can help facilitate set up, reduce stress, and avoid the need for troubleshooting when preparing for the performance. It is important to practice starting and stopping the piece just to ensure that if any problems are encountered during a performance, the performer is able to calmly stop and reset. Having a plan for technological failure is an asset.

Furthermore, it is good practice to rehearse electroacoustic pieces in full performance mode as many times as possible, instead of only learning the music first and familiarizing oneself with the electronics in the last few days before a performance. Practicing with the electronics as much as possible maximizes the chances of encountering pop-up notifications, alert sounds, and other issues during a rehearsal, allowing for ample time to avoid them in performance.

When placing a microphone to capture a saxophone, it is important to remember that the sound of the saxophone does not come entirely from the bell. It is recommended to use a condenser microphone (although ribbon and dynamic mics can also be used) placed between 6 inches and 2 feet away, pointing at the instrument at a height near the middle of the instrument. Experimentation with different distances can produce different tonal effects.

When moving a mic connected to an amplifier, disconnect it from the source of amplification or mute the signal by engaging a mute switch further down the signal path. Bumping the mic or mic stand may create loud sounds or damage equipment. There is also a possibility of generating audio feedback, which can be extremely loud and dangerous. Consider having someone next to the amplifier power switch or mixer when testing mic locations; they can silence the signal if necessary.

When connecting and disconnecting any cables, it is critical to turn off the amplifier before making or breaking any connections. Unplugging or plugging in cables with a powered

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amplifier connected can create loud noises that are extremely uncomfortable for listeners and may damage equipment.

**Always use the correct cables and connectors.** Just because a cable fits, does not mean it is an appropriate cable. Many of the cables and connectors used for electroacoustic music might be physically compatible but not *electrically* compatible. For example, most electric piano sustain pedals use a 1/4-inch TS cable. The sustain pedal cable will fit perfectly into the 1/4-inch input of an audio interface, but it will not work properly because it does not generate an audio signal. A common issue is using a pair of headphones with a built-in microphone that use a 1/8-inch TRRS male connector (Figure 16) to transmit a stereo signal (which uses a 1/8-inch TRS jack).

Although the headphones will fit into a 1/8-inch TRS jack, it may or may not work, or the sound may have distortion or problematic volume levels. The best solution is to use headphones that have only a TRS male connector.

**Avoid using adapters where possible.** Adapters may cause issues that are hard to diagnose. They also introduce additional points of failure and are easily lost or misplaced due to their small size. If adapters are necessary, try to find the highest-quality adapters, as cheap adapters often make poor physical connections and may introduce unwanted noise or distortion into the signal path.

**Make sure the performer can hear the electronics.** If the performer is unable to clearly hear the electronics, most mixers or amplifiers offer separate audio channels for the performer to directly monitor what is being played through the loudspeakers. If a system does not include an
onstage monitor (i.e., a loudspeaker placed on the floor pointing at the performer), then headphones will suffice.

**Whether using a phone for a performance or not, disable network connectivity.** For almost all electroacoustic performances, disable network connectivity by putting phones in airplane mode or turning them off entirely to avoid interruptions. Phone signals can interfere with audio signals, especially when near an amplifier, and can add unwanted artifacts and noise to a performance. Although using a phone for a performance, to play fixed media for example, is sometimes the only option, it is advisable to avoid using a phone wherever possible to limit unexpected problems during a live performance.

**When unsure about something,** communicate with the composer if possible. Many living composers are willing to share information about how they intended their piece to be performed and may have insights gained from past performances of their work.

2.10 Considerations for Specific Contingencies

**When performing pieces with fixed media that do not require specialized software or hardware,** the score will indicate how well-aligned the performer needs to be with the prerecorded track. Many pieces will have some or all of the fixed media part graphically notated in the score to indicate how the saxophone aligns with it. Others, however, are more improvisatory and do not need to be precisely aligned. Works such as *Saxpressivo* by Lawrence Moss have both improvisatory and precise moments. Moss writes in his work:

> Tape/Saxophone coordination is approximate unless otherwise noted (i.e., by [a vertical dotted line]). This is especially true during the long tape “improvisation” from the fifth stave of page 5 until page 7. The tape notation and timing are merely guidelines for the saxophone’s rhythmically free performance.\(^\text{41}\)

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Regardless of the situation, it is important to listen to and learn the audio track part as if it were a second performer. Approach these works with the same care and collaborative spirit as one would for a work with saxophone and piano.

**If using a computer to play fixed media,** a good option is to use a program called VLC Media Player. VLC Media Player is a free multimedia player that works on nearly every platform (e.g., Windows, Mac, Linux, iOS, Android) and offers extensive file compatibility.

**If using a phone to play fixed media,** disable the screen-locking feature and configure the screen to remain on during the performance. Some applications are set to use power-saving features when the phone is locked, which in rare instances can cause playback issues. Allowing the phone to stay on and illuminated throughout the duration of the work allows the track to be stopped quickly.

**If there is not enough time between starting a track and being prepared to play,** use an assistant to start the track. This can work especially well if the connection to the amplifier is off-stage. Another option is to find a silent track online\(^{42}\) or create a silent track in a digital audio workstation (DAW)\(^{43}\) and add it to a playlist before the performance track being used. When ready to begin, the silent track can be started, which will give a performer enough time to prepare before it automatically advances to the performance track.

**Have a plan for what is going to happen after the track ends.** If the performance track is part of a general library or playlist with other files in it, after the performance track completes, it could advance to the next track automatically. This often results in an unwanted track beginning while taking bows or walking off stage. The same unwanted result can occur if the

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\(^{43}\) DAWs are programs used for recording, editing and producing audio files.
playlist set to repeat. To avoid these mishaps, put the performance track into its own playlist and set it not to repeat.

**Avoid or delay updates and turn off notifications, sounds, and alarms.** Regardless of the operating system or the platform being used to play the audio track, pause system updates and disable notifications and other sounds, as these can interrupt the performance. Some operating systems have a Do Not Disturb function that will mute notifications. Additionally, avoid or delay system updates, especially if the update is requested between the last successful run and the performance. Updates, although critical to device security and functionality, can change the way programs act and can occasionally cause problems where there were none before. Additionally, disabling automatic updates or setting a time well after a performance for an update to occur will help ensure a computer does not automatically start updating (possibly causing it to restart) during a performance.

**When deciding on which foot pedal to use,** the physical noise that the pedal produces when depressed is a non-trivial factor. It can be distracting to hear loud or sharp clicks coming from a foot pedal, especially if this occurs during a quiet moment in the piece. Pedal volume should be part of equipment decisions, when possible.\(^\text{44}\)

**When performing with specialized hardware and software,** there may need to be additional setup required to reduce latency. Latency, or the time it takes for the audio signal to go from the microphone, through the computer and any requisite processing, and back out through the audio interface, can be affected by several factors (e.g., computer speed and audio driver

\[^{44}\text{Pedals that use metal switches like those on the BT200S-2 are generally louder than other foot pedal designs.}\]
settings). Although reducing latency can be complicated, latency severe enough where a piece becomes unplayable must be reduced.45

**When using USB cables to connect an audio interface or other device to a computer,** do not use USB cables that are longer than 16 feet for USB 2.0 devices or 9.5 feet for USB 3.0 devices. The cables will not operate reliably beyond these lengths. Additionally, whenever possible, avoid using USB hubs (a device that provides multiple additional USB ports). Some USB hubs, especially those of lower quality or which are unpowered, can struggle to provide connected devices with sufficient power to operate properly. This can cause intermittent connectivity issues and other unpredictable behavior.

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45 Directing programs to use Core Audio on Mac or downloading and using ASIO drivers on Windows can greatly decrease latency. Tutorials on how to do this can be found online.
Conclusion

Electroacoustic music offers exciting possibilities and rewarding experiences. This repertoire allows saxophonists to perform works that expose performers and audiences to a sonic range from the familiar to the celestial. With the emergence of newer technologies—in particular, computers and software programs—sonic possibilities and creative potential have never been more accessible. The vast variety of sounds and technology available for composers to use in their works, however, has led to a perception of the genre as complex and difficult. The real or imagined complexity of setups and knowledge required to operate hardware and software can be a significant enough barrier to alienate performers. The purpose of this guide is to help lower those barriers and create an understanding that encourages performance of electroacoustic saxophone music.

Through a dissection of electroacoustic works into their constituent parts, this guide provides necessary information that composers often omit. Many pieces exclude important details, often leaving it up to the knowledge and experience of the performer to come up with a solution for a successful and effective performance. With the information in this guide on understanding signal paths, making physical connections, common cable types used for those connections, live signal processing, and interactivity between electronics and performer, a performer can feel prepared to take the first steps of performing an electroacoustic works. Although the ideal method for conveying the information in this guide is through hands-on workshops and demonstrations, this guide, as well as the many links and references included herein, enables self-service, such that novice performers can learn enough to go out and try it themselves.
The preservation of a genre relies on continued performance and study. Electroacoustic works may not be as popular as other classical saxophone repertoire, but they are equally important. This genre represents an era in which the gap between technology and people began to rapidly shrink; understanding this genre is central to understanding the evolving musical zeitgeist of the 20th and 21st centuries. The technology needed to perform electroacoustic works has never been more affordable or available, opening the genre to performers usually drawn to more traditional, non-electroacoustic works. Performers today are further aided by innumerable online resources dedicated to electroacoustic repertoire, its performance, and the technology and processes behind it, many of which are free. This guide offers an often-overlooked piece of the electroacoustic puzzle, a sort of “user’s manual” for the electroacoustic saxophonist.
Bibliography


http://duramecho.com/Misc/SilentCd/.


https://www.worksforsaxophone.com/.


Riley, Terry. “Poppy Riley Poppy NoGood’s Phantom Band.” Program for *Solo Time Lag Music*


Further Reading


### APPENDIX A: Data Cables

<table>
<thead>
<tr>
<th>Cable Type</th>
<th>Description</th>
<th>Image</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>MIDI Cable</strong></td>
<td>Used to carry MIDI signals between MIDI capable devices. Identifiable by its 5 pins and notched enclosure. Although uncommon, they are still used to control a variety of older, MIDI only hardware. Both ends are identical.</td>
<td><img src="image1.png" alt="MIDI Cable" /></td>
</tr>
<tr>
<td><strong>USB A</strong></td>
<td>USB cables come with a variety of connectors, most often with USB A on one end. The cables below are used with a wide variety of devices including audio interfaces, MIDI controllers, keyboards, and many others.</td>
<td><img src="image2.png" alt="USB A" /></td>
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<tr>
<td><strong>USB B</strong></td>
<td></td>
<td><img src="image3.png" alt="USB B" /></td>
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<td><strong>USB C</strong></td>
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<td><img src="image4.png" alt="USB C" /></td>
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<td><strong>USB Micro B</strong></td>
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<td><strong>USB Mini B</strong></td>
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Table 1. Data cables