CHAPTER SAMPLER

Recording Techniques for Classical Music

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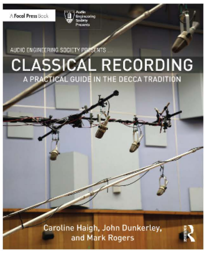
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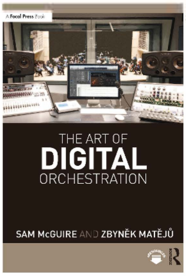
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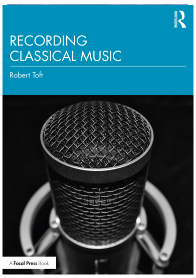
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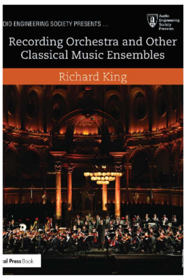
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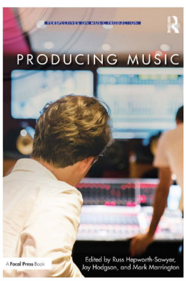
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Basic two-microphone stereo techniques

The focus in this chapter will be on the two-microphone stereo techniques that are frequently referred to throughout the book for non-orchestral recordings such as piano, chamber music, choirs, and organs. These fundamentally useful techniques are co-incident directional microphones, spaced omnis, and spaced and angled cardioids (of which the ORTF pair is the most well-known example.)

3.1 Co-incident microphone techniques

When two microphones are mounted as close together as possible (or when a dedicated stereo microphone with two capsules is used), the microphones are said to be co-incident, meaning that they are located at the same point in space. In theory, there are no timing differences between the two signals generated by such a pair, as sound waves from any source will arrive at both microphones at the same time. The resulting stereo image is produced by level differences between the two signals; these level differences are created naturally by the use of directional microphones pointing in different directions. Given that it is not really possible to mount two microphones in exactly the same place, there will be some very small timing (and therefore phase) differences, but these can be disregarded in practice.

To make the terminology clear, Figure 3.1 shows a general co-incident pair of unspecified directivity pattern (the patterns most used will be fig of 8, cardioid, and hyper-cardioid). The microphones are mounted with an angle between their front axes, and depending on this mounting angle and the directivity pattern, the pair will have a characteristic L-R 'stereophonic recording angle'. This is the angle between the positions of sources that will appear fully left and right in the stereo image produced by the pair, and it is not the same as the mounting angle. (It is assumed that the pair of microphones are fully panned.)

The L-R stereo recording angle is dictated by the level differences between the microphone signals that arise due to the directivity of the pair, and an individual instrument will only be perceived to be fully left or right when the level difference between its signal on each microphone reaches about 18 dB.[1] The width of the final image of the source will depend on how much of the pair's stereo recording angle it occupies. If the pair being used has a very wide stereo recording angle, and the source is a piano that only occupies a small segment of it, the recorded piano image will be narrow even with the pair fully panned. To make the piano wider without moving closer, the L-R stereo recording angle needs to be reduced so that the piano occupies more of it.

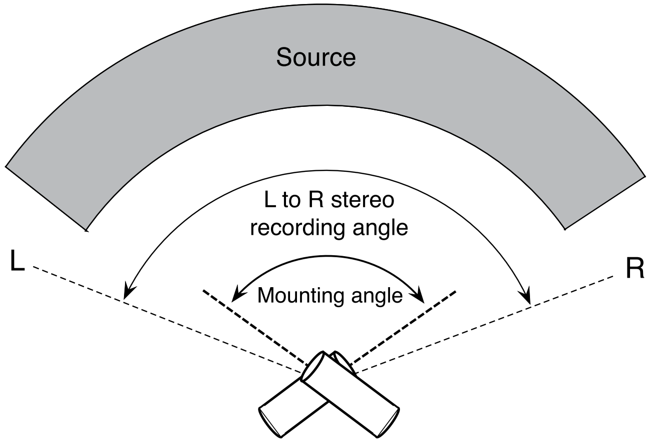

Figure 3.1 Generic co-incident pair

In general, an increase in the mounting angle results in a decrease of the stereo recording angle of the pair. This means that increasing the mounting angle will also increase the image width of the source.

The most well-documented co-incident microphone technique is the Blumlein pair, which consists of a pair of fig of 8 microphones mounted with their front axes 90° apart. This pair has a stereo recording angle that is also 90°, but care must be taken not to place any sources outside this angle to the sides of the pair, as the source will be picked up on the front lobe of one microphone and the rear lobe of the other, meaning that one signal will be phase inverted. When reproduced over left and right speakers or headphones, this phase inversion is a very unpleasant listening experience. Co-incident fig of 8 microphones are almost invariably used mounted at 90°.

When cardioids are used as a co-incident pair, the problem of a phase-inverted region does not arise, but some mounting angles will produce significant angular and level distortion in the stereo image. Angular distortion means that some parts of the image are compressed into a narrow arc, and other parts are stretched out to occupy a wide section of the image. This occurs because of the way that the level difference between the microphones changes with the angle of incidence. Level distortion means that some parts of the stereo image are louder than others (the areas of the image that are compressed into a narrow range are also louder.) The practical upshot is that cardioids mounted at 90° will produce an image which is centre heavy; most of the image is squashed into a narrow central range, with the rest stretched out at the sides. Conversely, cardioids mounted at 180° will produce an image that is stretched out and quieter across the centre and squashed up at the sides. If co-incident cardioids are to be used as a main pickup, they should be mounted at an angle of about 135° as this produces a useful L-R stereo recording angle and the least angular and level distortion. However, cardioids mounted at 90° can be useful for placing on a single, smaller instrument for use as an ancillary pickup. They will produce a very phase-coherent, centrally focussed image. Figure 3.2 shows the mounting and approximate stereo recording angles of cardioids at 90° and 135°.

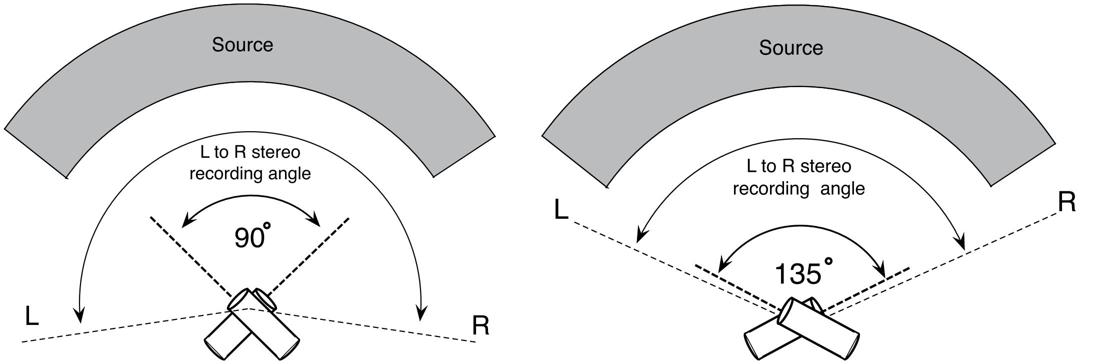

Figure 3.2a–b Cardioids mounted at 90° and 135°, respectively

Co-incident pairs create an image that is stable and coherent but lacks a feeling of spaciousness, and because directional microphones are used, the fullest LF range is not captured.

3.2 Spaced omnis

Using spaced omnis as an overall pair has both positive and negative characteristics. In its favour, the tonality will be excellent, with full LF range being included, and there will be a spacious feeling to the recording partly due to collection of more room sound, and partly because of the phase differences between the left and right channels. The downside is that the image will not be sharply focussed or precisely located, and if the microphones are too far apart, the centre of the recorded image will be low in level and the outer edges will dominate (the 'hole in the middle' effect). The lack of image precision can work to the engineer's favour if spot microphones are going to be extensively used; it means that they can be panned in a wider range of positions without causing conflict with the spaced omni pair. (See Chapter 13 for recording the classical brass ensemble as an example.)

The microphones will need to be mounted on a wide stereo bar (see Chapter 2), and it is common practice to angle them outwards at about 30° to 45° to take advantage of any HF directionality in the microphones. The stereo image when using spaced microphones as an overall pickup is predominantly created by timing differences between the two signals, although there will be some level differences at HF due to the HF directionality of omni microphones if they are angled outwards. The resultant source image width is therefore dependent on the spacing between the microphones, and this must be adjusted to take into account the size of the instrument or ensemble, its distance away, and the desired recorded image width. (See also section 5.5.2.) Spacing between fully panned omnis reaches a natural limit at about 1 m (3′4″), and beyond this, the timing differences start to become too large and there is not enough correlation between the microphone signals to create a stereo image. Reproduction of a large orchestra using spaced omnis is best addressed by use of the three-, four-, or five-microphone Decca Tree (see Chapter 8).

A small table of typical spacings used in practice to produce a conventional image width is included in Figure 3.3. These should be taken as a good starting point from which to adjust

Instrument/ensemble	*Guide to image width as percentage of whole*	*Microphone spacing*
Solo piano	50%–75%	25–33 cm (10″ to 13″)
String Quartet	50%–75%	60 cm (2′)
Chamber choir (20)	90%–100%	80–90 cm (2′7″ to 3′)
Chamber orchestra	90%–100%	80–90 cm (2′7″ to 3′)

Figure 3.3 Table of typical omnidirectional microphone spacings

position and spacing for image width and amount of direct sound versus reverb. To make the sound less reverberant, the microphones should be moved closer to the source, bearing in mind that this might also increase the image width by virtue of being closer to the players, so some microphone spacing adjustment might also be needed.

As noted in Chapter 5, spacing greater than about 33 cm (13″) on piano starts to produce an over-wide image that has poor localisation. Conversely, if a 30 cm (12″) spaced pair is used on an orchestra and fully panned, the image will still be too narrow to fill the space between the loudspeakers.

The microphones should be panned fully left and right (or very close to it). Panning a main pickup inwards to reduce the image width of the source has the effect of narrowing the sense of space and reverb around the instruments as well, and this produces a rather mono-sounding recording. Additionally, because this technique involves the creation of timing differences (and therefore phase differences) between the two microphones, panning them inwards to any significant degree (less than about 70% left and right) will start to produce some colouration due to partial summation of signals of different phase. When using spaced omnis as a main pickup (rather than for a 'stereo spot' as discussed in Chapters 6 and 7), it is best to space the microphones a little less if the image width needs to be reduced and to pan them fully.

3.3 Spaced and angled cardioids

A very useful technique that combines some aspects of co-incident pairs and spaced omnis is the use of cardioids that are spaced a small distance apart and angled away from each other. This general technique produces a lot of possible combinations of angling and spacing that might prove useful (in that they produce a useful L-R stereo recording angle and no significant angular distortion); these were quantified by Michael Williams[2] in 1984. The stereo image is created by a combination of level differences (due to the directionality of the microphones) and timing differences (due to the spacing between the microphones). The resultant image is more focussed than the spaced omnis, but with a more spacious feel than a co-incident pair. Where the extended LF range of omnis is not required (such as for a choir), they make a very useful overall pickup. As with the spaced omnis, the pair should be panned fully left and right as the default, and altering the angle or spacing would be preferable to panning them inwards beyond about 75% left and right.

Of all the possible spaced and angled pairs, two have passed into common use: the ORTF (Office de Radiodiffusion Télévision Française) and NOS (Nederlandsche Omroep Stichting) pairs, as shown in Figure 3.4.

It can be seen that the NOS pair has wider spacing and narrower angling than the ORTF pair, indicative of the trade-off between the two aspects of this sort of pair. As we saw in section 3.1

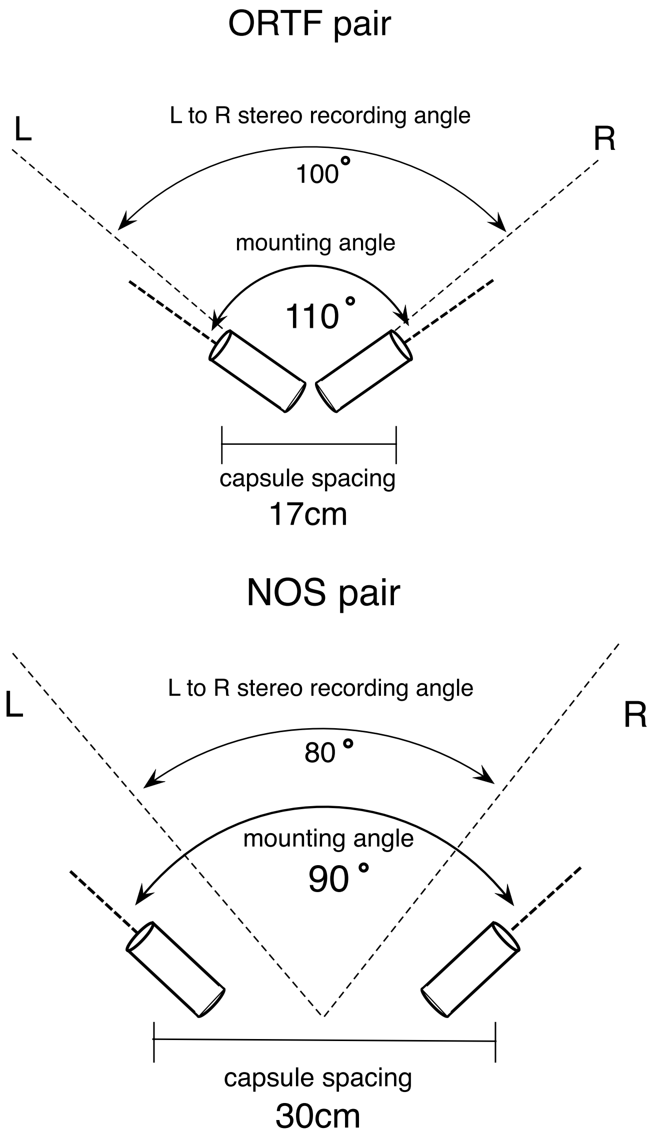

Figure 3.4a–b ORTF and NOS pairs

(co-incident pairs), a wider mounting angle means a wider image, and from section 3.2 (spaced omnis), wider spacing produces a wider image. The ORTF-type technique can be used flexibly in practice, and the angling or spacing can be altered a little to change the image width if changing the distance from the source is not an option. This might arise because the balance between direct and reverberant sound works well at a particular distance, or because the rig cannot be moved elsewhere because of space restrictions.

Elsewhere in this book, parallel-mounted pairs of spaced cardioids have been used extensively as spot microphones on soloists. This technique could be seen as part of the family of spaced and angled pairs as they have a mounting angle of 0° and spacing in the 20–30 cm (8″ to 12″) range. These 'stereo spot microphones' produce an image of the soloist that is wider than a single mono spot but is still appropriately restricted in width when fully panned.

Notes

1 See *Rumsey & McCormick: Sound And Recording*
(Focal Press ISBN-13: 978–0240521633)
7th Edition p498 for a useful summary of the research that underpins this finding.

2 *The Stereophonic Zoom: A Practical Approach to Determining the Characteristics of a Spaced Pair of Directional Microphones*
Author: Williams, Michael
AES Convention: 75 (March 1984) Paper Number: 2072
Publication Date: March 1, 1984
Subject: Studio Technology
Permalink: www.aes.org/e-lib/browse.cfm?elib=11692
There are interactive displays of stereo recording angles for various stereo techniques at www.sengpiel audio.com/Fragen08.htm

Orchestration stories and workflows

In this chapter, musicians and composers weigh in on specialized topics relating to the digital orchestration process. Each story was hand-picked in order to add to the narrative of this text. Not every point of view supports digital orchestration as the best option, and certainly some versions ended with a real orchestra recording the parts.

Sue Aston: digital orchestration

Composer and violinist Sue Aston has appeared on classical recordings, radio and television, both nationally and internationally, and worked with eminent musicians such as Simon Rattle, Nigel Kennedy, Peter Donohoe, Yehudi Menuhin, Sir Charles Groves, Esa Pekka Salonen, Gordon Giltrap and Chris De Burgh. She has also supported the folk legends Martin Carthy and Dave Swarbrick. The BBC1 *Heaven and Earth Show* presenter Simon Calder interviewed Sue about her composition work on one of their episodes about the Cornish landscape. Sue's solo videos have been broadcast on the Sky TV music channels Classic FM TV and OMusic.

Discography: Sue has released three solo albums *Sacred Landscapes*, *Inspirational Journey*, and *Between Worlds*, and a recent EP which features solo piano with strings and guitar – *Winter Keys* – and the DVD *Reflections of Cornwall*.

I use the Cornish landscape and nature to help compose my music. The stillness and background sounds of Cornwall weave through my music. Music can be the soundtrack of an ordinary day and an extraordinary day. The movies that play in our minds can conjure up their own soundtracks. Music can change the atmosphere of a room or a place or a frame of mind. Although I use folklore and legends and aspects of my own life to trigger each composition the instrumental aspect of my work is written so that the listener can weave their own images and experiences into each piece.

Sue Aston has also featured on the BBC Radio 4 documentary, *Derek Tangye: The Cornish Gardener* where she was interviewed by John McCarthy about her work. She has also been interviewed by BBC Radio 3 about her interest in English composers who were also influenced by the landscape. Radio 2 featured Sue's track *Initial Bond* for a program about composing music for a loved one who has died.

With advances in technology regarding the sampling of orchestral instruments, many composers use the sound of synthesized strings within the backdrop of a virtual orchestra when writing music. This obviously saves time and money. If the music score is busy, then the synthetic string sound can be very convincing, but in areas where the strings are exposed, especially during a solo section, then it is worth investing in hiring real string players to play over the track, to add an authentic performance.

In particular, certain aspects of string performance and technique reveal whether or not a piece of music features real string players. Different registers of a string instrument can sound thin, especially at the extreme high and low end of the range. In combination with the use of vibrato, which is often too narrow and fast, this can really highlight the fact that the sound is produced electronically. During the performance of a single note, a string player will create a multitude of different sounds by changing the bow speed and varying the vibrato. The sound of the note will grow, by building up to a crescendo, then fading away with a diminuendo.

Synthesized strings don't offer the scope needed for sensitive playing with the bow, and techniques such as *flautando*, *sul tasto*, *sul ponticello* and creating natural and artificial harmonics can all sound contrived. Pizzicato can sound too harsh, snap pizzicato and *col legno* can sound far too percussive. Tremolo can especially sound too rapid and unconvincing.

The very aspects which make a string instrument sound beautiful are missing – in a real performance the notes are created by the player carefully balancing the bow with an even weight between the fingers, then bowing the hair over the strings, which in turn vibrate and manifest the sound out of the depths of the varnished wood.

From my own perspective, when I have been asked to play in a recording studio over an electronically produced track, factors to take into careful consideration when performing are that a cold wooden instrument may take time to warm up, and special attention is required regarding intonation.

String players traditionally tune by ear and listen to the interval of a perfect fifth, whereas a synthetic string sound will be too bright and artificial sounding. Vibrato needs to be used mindfully, as if it is too wide, this can alter the pitch slightly and clash with the perfect tuning of the electronically produced music. Playing to a click track doesn't allow for rubato, and so it is difficult for the music to naturally breathe and flow.

Despite these differences and challenges, using real string players enhances any music score, by adding a depth and rich quality of sound. You may assume that you would need to employ a whole string section to achieve this result, but I am often hired to play and record multiple layers of violin and viola parts.

Real strings add an authenticity to what can sometimes be a one-dimensional, flat sounding layer of electronic music. It creates a perfect balance between the use of traditional string instruments and modern musical recording techniques.

Col. Vaclav Blahunek: wind symphony orchestration

Col. Blahunek is the Director and Chief Conductor of the Prague Castle Guard and Czech Police Symphonic Band.

1. What should a composer know before they compose music for a wind orchestra?

I would expect these absolutely necessary skills of composers for wind orchestra. They should know:

- Tone ranges and sound possibilities of every instrument.
- Dynamic scales of every instrumental section.
- Technical difficulties of wind instruments, like breathing, slurring, articulation, attacks etc.
- Forms of music.

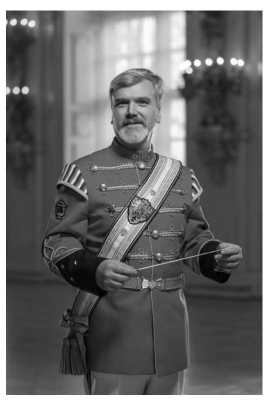

Figure 7.1 Col. Vaclav Blahunek, Ph.D.

- Rules of counterpoint.
- Concepts of musical styles.
- To know "how to score music for winds" = experiences and research into wind band scores.

Special features:

- Unique compositional techniques.
- Exploring new sound qualities and expression of winds.
- Easily recognizable personal style within almost every measure of music.
- Magic of musical speech: "to know what to say and how to say" in which case understanding of listener makes no difference for it!

2. What mistakes do composers make when writing for a wind ensemble?

The most frequent mistakes of wind band composers:

- Lack of respect in the balance of sound (it is usually the main responsibility and creativity of a conductor).
- The blend doesn't work.
- Too many harmonic mistakes (sometimes typos due to the publisher).
- Missing respect of special wind characteristics of overtones.
- Pieces of music are too long – they do not know where and how to end the piece.

3. What mistakes do arrangers make when transcribing music for a wind ensemble?

Mistakes of arrangers of symphonic scores for wind ensemble include:

- They do not respect tone ranges.
- Patterns of process, e.g. strings to clarinets, cellos to saxophones.
- Does not respect harmonic progress of composition, they often change notes!

The best way to orchestrate music is to "decompose" the piece, find the main structure, uncover the roots and then match it to totally different sound colors, suited to the wind medium.

Don Bowyer: composition/orchestration process

Prof. Don Bowyer is Dean of the School of Arts at Sunway University in Malaysia. He has played trombone in more than 50 countries and has published more than 60 compositions that have been performed around the world.

Composition biography/use of technology

As a composer, almost all of my work has been for a specific group of live performers. As such, my primary technology usage has been notation software and a MIDI controller, focused on printed notation much more than audio output. I should also mention that I am an active performer on trombone, which I suspect has had an influence on my compositional style and process.

I started composing in the late 1970s, using pencil and paper, later copied in ink. I do not miss those days, although there was a period in the 1980s when I had a decent "side gig" as a copyist. When Finale 1.0 came out in 1988, I got very excited and decided I needed to switch to computer-based composition. While I was (and still am) a Windows user, the first version of Finale was Mac-only. Fortunately, I had a Mac SE in my office. By the time I left that job two years later, I had switched to the Windows version of Finale.

I still remember being intimidated by the unboxing of that first version of Finale: three spiral-bound instruction manuals, each an inch or more thick. I was just OCD enough to actually read them all the way through, but it still took years to feel proficient. My other clear memory from those first days was w-a-i-t-i-n-g for the screen to redraw. It literally took several minutes, sometimes even ten or more, to redraw the screen for a full score on that Mac SE. In the early days, we learned to edit without redrawing the screen, then redraw after several edits. Of course, that meant remembering what changes you made in one part as you edited another.

I did not have a MIDI controller in the early years. Even so, I quickly discovered that Finale's Speedy Note Entry was a much faster method of note input for me than the Simple Note Entry. I now use a MIDI controller for almost all input, but still with Speedy Note Entry. I have had several miniature controllers I liked, recently including the 25-key XKey and the Bluetooth-enabled NanoKey Studio. My computers have all been laptops for at least ten years, so the smaller keyboards enhance that portability.

Because my ultimate goal is typically printed music, I primarily use playback only to identify wrong notes, as in "I don't think that was what I wanted." As such, I have mostly used the built-in sounds. Recently, however, I upgraded to Note Performer because the Covid-19 lockdown forced me to perform a few times with Finale files. I like the string and piano sounds a lot more now. On the other hand, after performing through Facebook Live, delivered over phone speakers, I'm not sure anyone else noticed!

My earliest compositions were almost entirely for jazz groups – both big bands and combos with two or more horns. In more recent years, nearly 50% of my composing is for "classical" soloists or ensembles, both large and chamber.

Orchestration process

Besides composing, I have also done a fair amount of arranging or orchestration (which I see as the same thing, though I know that some do not). For both jazz and classical work, my compositions usually start with melodies or, occasionally, a rhythmic motive. The harmony almost always comes later in my process. In other words, I usually compose horizontally, then orchestrate vertically. As such, the orchestration work for me is pretty much the same whether I am composing my own work or arranging an existing work. To be clear, I am not advocating for this approach – I suspect most of my favorite composers have a different process that blends the harmony and the melody into a more symbiotic whole. For me, however, single-note melodies come more easily. Perhaps this is influenced by my years as a trombone player (and my lack of keyboard skills).

Jazz orchestration

Most of my jazz compositions are for big band. As a graduate student I had an arranging class with a fellow graduate student, Dan Gailey, now the long-time jazz director at the University of Kansas. Among other things, he shared with me a voicing chart, based on Basie voicings, that he said he got from Tom Kubis. The chart included suggested four- and five-part voicings, both open and closed, for every scale tone in major, minor, dominant, diminished, and altered chords. In my early days, I spent a lot of time with this chart, meticulously duplicating the Basie sound. Of course, I eventually internalized this enough to "break the rules" and develop my own style. I no longer know where that chart is, but I am sure that I still use the same sounds more often than not. Applying this, if I have a melody in the lead alto, the other four saxophones flow down from that. Then I go back and look at voice-leading, particularly repeated notes in an inner part, and make adjustments to smooth the lines. The same process holds for each horn section. When combining sections, I use a couple of different approaches. Trombones, for example, might duplicate the trumpets an octave lower. More often, though, in a section with all brass, with or without saxes, I will try to create an interesting bass trombone line instead, filling in the horns between using the same four- or five-part principles, but spread over eight or ten brass.

I should mention that I also use a fair amount of unison lines in horn sections, sometimes with all horns, and sometimes only within the section. One sound I possibly use too much is writing simultaneous independent unison lines for each section.

My approach to the rhythm section probably grows out of my background as a horn player. For almost every band I write for, I believe the bass player will come up with a better bass line than I will write. The only exception to this would be a specific repetitive bass line that I might want. Likewise, I believe the keyboard and guitar players will provide better voicings than I will, so I usually write chord charts for them. Incidentally, I rarely write rhythm parts so I can hear them in playback.

For drum parts, I will indicate style and tempo, and when to play time. If there are specific hits I want the drummer to play, I will write those in the staff (the rhythms, not the specific instruments). The last thing I do with the drum part is to write in horn cues for most of the chart. I put this in a separate layer in Finale, above the staff. I create the cues by copying the rhythm of the lead horn at a given moment. When writing for smaller jazz groups, I use the same principles as the big band, but with fewer horns.

In summary, my approach to jazz composition and orchestration is performer-informed. Jazz has been primarily a performer-based art form. I consider the relationship between composer and performer to be a collaboration.

Classical orchestration

Whether because of my jazz origins or my performance career, I have the same collaborative approach to classical composition and orchestration. I don't write improvised percussion parts (unless it is an aleatoric piece), but I do welcome performer interpretations that don't match my original conception.

My compositional approach in classical music tends to vary from my approach in jazz, particularly in chamber music, where each instrument will be more independent and, therefore, less "orchestrated." My large ensemble pieces, though, have a similar orchestration approach to my jazz process. The harmonies and voicings are usually different, but the sectional approach is similar.

Additional uses of technology in my compositions

As an aside, I have become very interested in recent years in aleatoric music that involves the audience in some way. This is not really an orchestration process, but it does bring up a couple of other examples of use of technology.

One piece that I have performed fairly often is for trombone, multimedia, and audience cell phones. I wanted to involve the audience in some meaningful way in a piece that serves as a metaphor for a day in my life. I composed individual sections of music for trombone and electronic sounds. The electronic sounds are in a Flash timeline (not a recent composition) that includes animations projected on a screen. The individual sections are titled for activities in my typical day: administration, composition, performance, technology, teaching, and family. A seventh section, called unplanned demands, interrupts the music every time my cellphone rings. Following a short interruption, the music picks up where it left off. My cell number is always visible on the projector screen and the audience is encouraged to call as often as they "need to."

In other technology uses, I have composed a couple of pieces for instruments involving a looping app on my phone. Finally, I have composed a couple of small film projects, synchronized through Ableton.

Tip for writer's block

In closing, I thought I would provide a tip for overcoming writer's block. When I am struggling to get started, I tell myself to write one note. Then write one more. Repeat until something sounds good, then go back and delete (or edit) the parts that don't sound good. This process is so much easier with notation software than it ever was on paper!

Rahul Shah: library analysis

Rahul Shah is a Canadian composer for film, television and multimedia from Toronto, Canada. His sound has often been described as classical/orchestral crossover with world elements. Rahul has written music for projects distributed on Netflix, Amazon, PBS, BBC, ITV, Discovery and worked on advertisements. He has gained professional experience in composition, orchestration, editing, score preparation, sound design, and arranging. In addition to his own freelance projects, He has assisted composers for film and television at Hans Zimmer's Remote Control Productions, including Henry Jackman (*X-Men*, *Captain America*, *The Interview*, *Jack Reacher*, *Kingsman*).

There is no shortage of sample library developers on the market, just naming a few we have VSL, Spitfire, East West, Orchestral Tools, Cinematic Studio, and Cinesamples. All of these libraries and sample developers pose advantages and disadvantages to their software, but sample libraries have come extremely far in the last 15 years. Much like cell phones just the fact that a smartphone exists is conceptually amazing. I've heard stories of composers from 20–30 years ago running sample libraries off of floppy disks and other storage mediums far more primitive in comparison to the hard drives we use today.

Let me be clear, I appreciate many different libraries from all of these sample developers. They do great work and deserve for you to buy their libraries. In saying that, I'll share some of the comments I've heard in the industry about certain libraries. People have argued in the past that Spitfire, while it sounds great out of the box isn't always the best representation due to limited mic positions, articulations and a lack in dynamics. Some have argued that Spitfire is a very good, out of the box, solution but prior to BBC Orchestra it didn't provide as realistic an implementation of an orchestra. VSL (Synchron) has been mentioned, much like East West's PLAY engine, to be somewhat heavy on the CPU and RAM resources of your machine. Orchestral Tools features stud libraries such as Berlin Woodwinds and that is definitely my favorite choice for woodwinds (they also have dynamics such as *m*, *ff*, *f*, *p*, etc.) but there is no further control on quantity. Cinesamples does a great job recording, editing and managing samples of their libraries. However, if you're not going for a large sound they can be unnecessary for certain arrangements. With all of these libraries there is a reason why people still pay thousands and sometimes millions of dollars to record with the top orchestras in the world! You always need certain human elements (natural human imperfections in playing, breathing, air, the room), which create the true feel of an ensemble.

I'm sure everyone uses real instruments, however, for demo/mock-up purposes getting an accurate representation of your music is important (e.g., dynamics, articulations and feeling as close to a live performance). There are situations where a live ensemble/player isn't available or an option – for this reason, I really like the libraries from VSL (Vienna Symphonic Libraries) if that is your main goal.

Their libraries can also serve you well in a piece written for concert. For the most part they have the dynamics and articulations you'd need and the samples are recorded well. Top educational institutions in composition or film scoring recommend or provide VSL as a bundle to learn command of an orchestra and hear realistic implementation of what you've written. VSL does a great job of nailing the exact orchestration and timbres (e.g., three horns sound like three horns) rather than only having access to a solo horn, or two horns or 6/12-horn patch. VSL has done a great job of making sure that they are placed well in the room so that if managed properly, instruments or sections can provide the illusion of true orchestral panning/ensemble. To translate this, it sounds like your music is being playing by humans and after you sequence the patch in your DAW (e.g., Cubase, Logic, DP, Pro Tools) they're realistic and sound like a human could've played them.

For a composer who wants to have the most realistic simulation of what they're writing so they can hear what their ensemble will sound like, Vienna Instruments/Synchron-ized series (Volumes 1, 2, 3, 4) provide the most accurate picture to them. When orchestrating it's extremely important to know what the weight of your ensemble is, you don't want to use a 6 horn or a 12-horn patch when your ensemble has two horns because you'll wonder why your horn sound isn't as strong. When you combine the exact dynamic references that are available in Vienna (e.g., *mp*, *f*, *p*, *ff*, *fff*) this going a step further to make sure you are hearing exactly what you or someone else has notated in a score. Similarly, if you have two clarinets and two flutes but you're using a patch with five to ten woodwinds each your expectations will not be met during a live recording. Additional parameters can be programmed in Vienna Synchron player such as Xfade (Velocity Crossfade or other MIDI CC (control change) parameters. A big problem with orchestrating for strings has to do with numbers, these string patches in some libraries can have 10–12 violins in one violin patch. If you add a whole family by section you could end up expecting the sound of 50+ string players and in the actual recording you may only have 15. In terms of hybrid orchestral, because the sounds are so realistic when merged with live players from an original recording it forms a hybrid sound. Then if you need additional thickening of the orchestration for a purpose then one could add a bit of extra weight from the other libraries to thicken once the recording has been completed. This will need to be evaluated according to the context, an intimate chamber sound isn't likely to require as much thickening as an orchestral epic trailer or action scene. Not to mention there are countless instruments that are used in concert writing that aren't even sampled with the other sample developers just based on the fact that their prospective buyers wouldn't generally have use for these instruments in cinematic writing. Having these additional tools either for scoring or writing is definitely an added bonus over other sample developers and libraries.

Here are a few examples from the concert world and film scoring world where I exemplify how Vienna translates well from score, demo/mockup, live recording and then to the final dub stage. You can take the score from any films or concert, program it into your DAW of choice using MIDI and you're likely to notice Vienna Instruments/ Synchronized libraries provide the most realistic implementation of what you see on the score. The best-case scenario is getting a realistic sound with true illusion of ensemble as well as an epic sound that fits the context after thickening with other libraries to create a sample hybrid and hybrid orchestration along with a live recording. At Berklee College of Music one of the assignments I remember in the scoring program is sequencing these pieces/classic works to create mockups and create the illusion of an ensemble using MIDI.

Figure 7.2 Excerpt from Pavane pour une infante défunte by orchestrator and composer Maurice Ravel

For instance, this excerpt by Maurice Ravel requires sounding like two horns – Vienna libraries do that and also allow you to select the *pp* dynamic in the patch. It is also important to not quantize the melody 100 percent according to a grid because then it doesn't feel as musical as the original piece. Allow for human pacing so that the integrity and phrasing of the piece is maintained. You can then rely on MIDI CC 1 (Modulation) to add some more expressiveness to the phrase and further shape the line. Samples are recorded for each dynamic layer (*pp*, *mf*, *f*, *ff*, *fff*, etc.) instead of being duplicated. It's important not to just use a two-horn patch but to use a solo patch as a separate performance to create the illusion of a small horn ensemble

This excerpt by Copland is a great example of how Vienna libraries can be used effectively for an orchestral piece while maintaining the sound of the brass section (four horns, three trumpets, three trombones, and tuba). You many need to layer using more powerful libraries for the desires sound but if your goal is to create exactly what is written on the page in terms of number of instruments, dynamic layers, realistic attacks/releases, human dynamics and articulations (e.g., *marcato*, *staccato*, etc.) then I think Vienna is a great place to start. It is also a great learning tool to hear how various numbers of an orchestra sound so that you aren't as surprised during the recording session when it comes to the perceived weight of your sound. This can also result in you orchestrating differently and approaching registers of the respective instruments in a different manner.

Ondrej Urban

Ondrej is head of the HAMU Sound Studio and the Department of Sound Design at HAMU

My preferred way of orchestral recording is to understand all of the components of the process: score, orchestra, hall. The most common practice is to build the basic re-

Figure 7.3 Handwritten excerpt from Fanfare for the Common Man by composer Aaron Copland https://loc.getarchive.net/media/fanfare-for-the-common-man-1

cording system around few mics at the beginning. It really depends on wishes and needs of the final recording - I mean, I can choose different setup for soundtrack movie and another one for making just sound recording for archive purposes, for radio or TV.

Starting with the basic stereo-mics setup really helps to understand what the sound of the hall is, projected to a basic system. I usually use ORTF, XY, AB or Decca 3 setup or combination of them, depending on the size of stage and orchestra. If the score is written well and the orchestra is playing well too, half of the job is done. Then, after listening to the basic stereo setup, I ask, what is missing. Often, I have musical director by my side, or I have the score for myself and I need to "read" it before recording. The missing components are usually the direct bass signals, presence of solo instruments, attacks of drums and percussions. Making the basses less ambient helps to build the sound sitting firmly on the bottom. Spot mic for solo instruments or orchestral solos (e.g., flute) helps for putting the solo voice in front of the filed, more "in your face." And spot mics for drums and percussions (mainly timpani and mallets) improve the attacks and perfect rhythm pulses. Another spot mic helps the harp to be more present, also horns will like it. So, after some adjusting, we have basic stereo setup plus a few spots, circa 12–20 tracks. My idea is to make the basic sound as good as possible only using stereo system and then to help with some spots. This approach works well with more classical scores (Beethoven, Dvořák, Brahms…) and with sound for radio and TV.

Another way of building the sound field is recording for movies, pop-oriented orchestral recordings etc. I go usually in the opposite way: I start with separate and more directional mics for groups of instruments – 1st violins, 2st violins, violas, cellos, basses – than woodwinds (two to four mics), brass and horns (two to four mics), percussion (two to five mics), soloists (one to three mics). Sometimes we have organ, choir and for example orchestral piano, celesta … (so, a few mics more). I prefer more contact miking in such situations, so as to have more of the presence. But for "gluing" everything together, I use a combination of ambient mics (mainly AB) and artificial reverberation. The final sound is more "in-your-face," you are not limited by the dynamic levels of different orchestral instruments and their groups etc. It is this way that I usually do the soundtracks for movies or computer games. Depending on the needs and size of the orchestra, we record usually in the studio (drier) or concert hall (wetter). My favorite concert halls in Prague, where I do the orchestral recordings are: Dvořák's Hall of Rudolfinum, Smetana's Hall of Municipal House, Martinů's Hall of Lichtenstein Palace (Academy of Performing Arts).

Figure 7.4 The Rudolfinum Concert Hall

In movie soundtrack recordings, pop-music recordings and recordings for games we usually need to sync with the picture and/or pre-recorded material (sampled instruments, rock band recording, click etc.) In such a case, every player needs to wear headphones. The main difficulty is the cross-talk from players' headphones to each section mic. I usually adjust the level of click and playback tracks to be as low as possible, giving the players the right tempo, but not having too much bleed.

Camille De Carvalho

I only use real instruments when recording music. The reason goes back to when I was 16 – at that time I was just a young pianist and I had just bought an electric keyboard, wanting to connect it to my PC to play new sounds. I installed notation software and began writing with all the sounds available, which I had mostly never heard before. My family is not really into music so apart from piano, guitar, drums and maybe saxophone I didn't really know any other instruments. I also installed a lot of keyboard emulations: Minimoog, Hammond organ, Fender Rhodes … and I was quickly overwhelmed by the quantity of different sounds I could play. How could I ever choose between them?

My pieces were not progressing at all as I was constantly tweaking the knobs of my virtual Minimoog and rerecording everything. And then I met Stephane. He saw me playing piano in high school and introduced himself as a flutist. At first, I laughed at him: what a useless instrument! Monophonic? Only three octaves range? Why not playing piano?

"Not the same sound," he said.

I answered that I could do every sound I wanted with my computer.

"Oh yeah? Well then, let's try." He sent me a short recording not even using strange techniques of him playing flute and asked me to replicate it. I tried, I struggled with vibrato, with dynamics, with the erratic tempo, with the breath sound – I couldn't.

Then he lent me one of his old flutes and told me to try it, explaining the basics to me. I discovered that there is more to music than pressing keys. Finally, he invited me to a rehearsal of his orchestra. That's when I understood that the infinity of sounds you can make with a computer does not include the infinity of sounds you can make with real instruments. So I had to choose a side: I could stay with my computer and use these sounds, but I already knew I had trouble picking one sound, or I could try with real instruments where you have a limited number of instruments, a limited number of techniques, to see if it was better for me.

So, I began writing orchestral pieces but, as I had no orchestral culture, my pieces were unusual to say the least. Not a lot of place for strings, a lot of rare winds and a lot of parts were actually unplayable. I also found out that if you're not already known, no orchestra will even bother looking at your pieces and if you can't pay session musicians, you'll have a lot of trouble finding enough of them to record an orchestral piece. I decided to do it all by myself. Luckily, my parents moved to China when I was 18, so the first time I visited them, I headed to Music Street where you have musical instrument shops for miles and bought my first woodwind instrument, a clarinet.

Since then, I have spent a lot of money on instruments and lessons. I bought all the members of the woodwind family I could find, then went on with the brass family, then the strings. I took lessons for some of them and used the fact that there are a lot of similarities to learn the others. I think the best bet is to take lessons for the hardest instruments to play: oboe, horn and violin. Then you can play most of the instruments in the orchestra. Of course, I'm not an expert in each of these instruments and sometimes, I need to write some difficult parts I can't play. But ten years in different orchestras helped me meet a lot of people and it's really different to ask a violinist to play 16 parts of long notes during 15 minutes than to ask him to only play the nice solo part. But the most important thing is that while learning how to play these instruments, I learned what is idiomatic and what isn't. Nowadays, it's true that some libraries can emulate string instruments really realistically and you can approach most sounds (although still today, I never found any piano VST with a mute. I love the mute on upright pianos!). But something that is still noticeable is when you hear something an instrument isn't supposed to be able to play. Even if you're not a musician, it sounds wrong. I once heard a chromatic harp glissando for example: the sound almost convinced me, but I knew it wasn't possible (although some chromatic harps exist, but they're really rare). And learning all of this taught me how to write so it sounds legit. Also, I can't change my mind at the last moment and replace the oboe sound by a clarinet sound: every note I write must have a meaning, a reason, or I'll spend hours recording for nothing. Every time I listen to one of my pieces, I can say proudly say "I recorded these flutes one by one, little by little," even though even a mellotron sound would have done the trick. The satisfaction I get out of it is immeasurable.

Cory Cullinan, a.k.a Doctor Noize

Digital orchestration is not always about making the final project digital. There are times when I use digital instruments to actually render and perform the final recording of my arrangements for both Doctor Noize and other projects. But there are also still many times when you go with "the real thing" in the final recording. It's important to note, however, that almost nobody goes from conception to recording a large acoustic ensemble without digital technology anymore – in either the arranging or recording phase. Hats off to those brilliant purists who do, but honestly, even a genius like Mozart would probably use digital technology in almost every phase today. Most music copyists now work via digital means instead of by hand, too

There are many reasons for this. One of the main ones is that advanced scoring software allows you to extract parts from a full score instead of writing them all out again for each instrument. If you've never scored for orchestra yourself but have seen the film *Amadeus* you know how helpful this can be. In the film, Mozart is struggling against health and timelines to finish his *Requiem*. It's not the composing that's slowing him down; it's the time it takes to get his brilliant creation onto playable scores and parts for the musicians that's keeping him up at night and killing him.

Anyone who's ever written for a large ensemble – orchestral, choral, musical theater – can relate to this, even if it takes us longer to compose than it took Mozart. Mozart could write complex music straight from his head to the page (it was like he was taking dictation from God, said his rival in the film), but no amount of genius could reduce the time it took to then copy each instrument's part from the full score onto a new page. That was just grunt work.

Figure 7.5 Doctor Noize

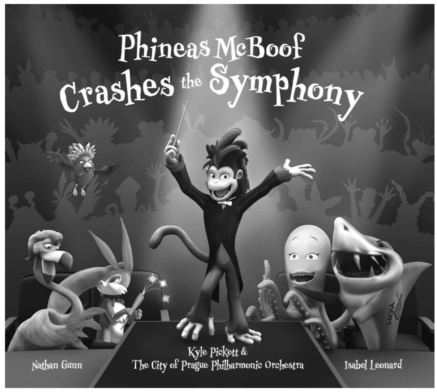

Figure 7.6 Album Cover

Advanced scoring programs like Sibelius and Finale take much of the grunting out of the grunt work. More than ever, with improvements over the last decade from software developers, full scores can be made into parts extremely quickly in a process called extraction. Most of us compose in full score mode, and then extract those parts into individual scores for violin, bassoon, bagpipes with distortion pedal, and any other parts you've written. (Okay, fine, I've never scored a piece for bagpipes with distortion pedal – but that's my loss and the world's, and frankly a gap in my production output that begs to be rectified.)

In my experience, it's a marketing myth that professional-quality parts can be extracted with the touch of a button in these programs. Almost always, several things that were laid out well in the full score don't transfer over perfectly – a dynamic too close to a note, a text box that now sits in the wrong place, or even a problematic page-turn for the player. But instead of taking dozens of minutes or hours to copy the part by hand from the score, you can generally make the proper adjustments to an extracted part in just a few minutes

A glance at this page from my Doctor Noize double album *Phineas McBoof Crashes the Symphony* shows just how much time and life expectancy can be salvaged from these gains. Mozart would be jealous of the resources at my and your disposal.

Figure 7.7 In the Studio

Figure 7.8 Live Performance

This project, which I recorded with the City of Prague Philharmonic Orchestra and 20 vocal parts (16 solo roles and the four-section Stanford Chamber Chorale) was a doozy to produce. Do the math in your head: The work is two hours and fifteen minutes long, with 20 vocal parts and over 30 instrumental parts. The full score is hundreds of giant pages – and that's *before* I had to extract all the parts – and contains dozens and dozens of musical numbers and scenes. Just writing that, or looking at my beautiful completed score now, makes the modern mind think: "No way I'm ever doing that by hand."

Even with this technology, it was a gamble for me on every level – financially, commercially, artistically. After a well-received tour of Doctor Noize orchestral shows introducing kids to the orchestra – started by an inspired commission I received from the McConnell Foundation to write a live orchestral show for kids, families and schools – we made the commercially questionable decision to make it into a recorded work so kids, families and educators who couldn't attend a show could still experience it. And we decided to make it in two full narrative acts, to give kids the experience of a legit opera, because we thought modern kids and even modern adults were smart. (Every night as I looked in the mirror working on this project, I realized that second assumption was a big leap …)

We had secured just enough funding to produce it through a digital fundraising campaign on Kickstarter that was ultimately profiled in detail in the European Business Review – we had a major orchestra and Grammy-winning opera stars Isabel Leonard and Nathan Gunn had committed along with our normal Doctor Noize cast. But two things that haven't changed since Mozart's day are the unknowns of whether the production will go off without a hitch and whether or not the public will like it.

Figure 7.9 The Dream Team

When you imagine that composers like Mozart and Wagner composed operas even longer than *Phineas McBoof Crashes the Symphony* with no digital technology whatsoever, and with their careers, reputations and livelihoods on the line, you suddenly gain an even deeper appreciation for the work of these masters. The production took me over a year of fairly constant work to compose and produce. But unlike the aforementioned masters, I had digital orchestrating software that allowed me to audition the ideas that came out of my head to hear if they sucked and change them before anyone else heard them; use quickly-exported digital representations of the score along the way in vocal rehearsals instead of piano reductions; demonstrate at any time to a potential performer, funder or future commissioner just how much work I had done, in pristine and beautiful score form; and, equally important for a composer's state of mind, easily create digital copies in multiple locations as backups. Imagine being two thirds of the way through writing Beethoven's Ninth or Mahler's Eighth after years of work, and going to bed every night knowing that the only copy that existed in the whole world could be burned down in a building fire at any time! It's enough stress to make a composer like Mozart die before he hit 40. Wait a minute …

So. Even with all the best digital technology in the world at my disposal, I *still* almost screwed it up. I will be the first to admit that digital scoring software is what bailed me out. It literally was the pivotal difference between a disastrously incomplete overly-ambitious failure and a glowing success story with many accolades that somebody just invited me to write a book chapter about. Here's how.

Figure 7.10 Trying to Stay Awake

Figure 7.11 Excerpt

Despite the fact that I had been down in my production facility Reach Studios until 1am every night for months toiling away at the digital scores, awakening at 6am to get back to work; despite the fact that my wife was convinced I had become like Richard Dreyfus in *Close Encounters of the Third Kind*, Captain Ahab in Moby Dick, or Mozart with far less talent writing his Requiem in *Amadeus*; despite the fact that I had raised and accepted over $100,000 from generous music-loving fans and donors who wanted to see this work completed; and despite the fact that I had planned production and the recording phase over a year in advance … I boarded a plane in Denver scheduled to arrive in Prague about 15 hours later sleep-deprived and with four critical full scenes still yet to compose and arrange for the orchestra I had just committed $45,000 to record the project. The music was in my head, but it was not in score form

I worked the entire set of flights on the scores. I quickly realized it was far more laborious work than at Reach Studios because instead of my 88-key weighted keyboard to enter notes into the score, I had to grab and place the notes by hand with my trackpad. Nothing to do at that point but work harder. I did not have noise-cancelling headphones, which was brutal. (I have never boarded a flight without noise-canceling headphones since.) I found a plug somewhere else on the plane and would take nap breaks to plug my computer in for a while. (Thankfully nobody can leave the plane halfway over the ocean if they steal your stuff, but still a desperate and dumb move.) When I landed, I had *two* of the four final scenes banged out – conductor's score and parts. But two still remained, and I was so tired I could hardly think

Figure 7.12 Excerpt

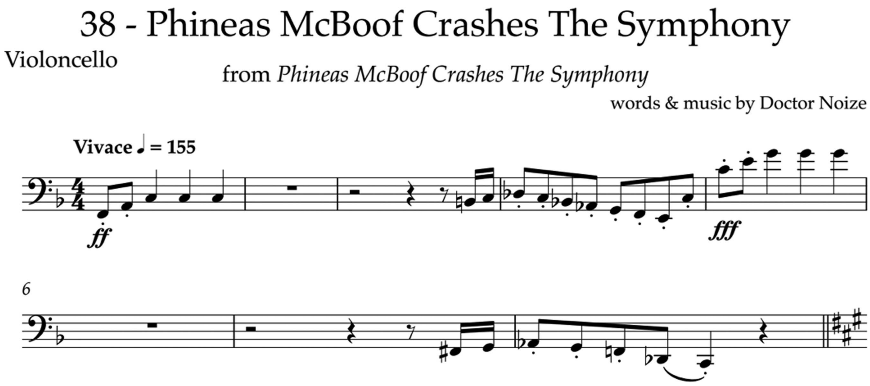

Figure 7.13 Excerpt

I got to my hotel, emailed the latest scores to the orchestra's copyist, and tried to sleep. But I couldn't, knowing I had not completed the scores but had accepted everyone's money for the project. So, I got up and finished one more that night instead of sleeping at all. When I showed up that first day with the City of Prague Philharmonic Orchestra, I probably looked like a corpse. Thankfully, I had flown my favorite conductor, Kyle Pickett, in with me. Kyle and I had known each other forever; we were Stanford music students together, and he had conducted our tour of orchestral shows. He knew much of the music already, and I trusted that he understood my instincts if I was brain-dead at the recording session.

It was comforting that Kyle was there, because I had never met anyone in Prague before that day, and I was pretty sure the orchestra's musical director James Fitzpatrick thought I was an idiot American because I had called the Czech Republic "Czechoslovakia" numerous times in pre-production email exchanges, even though the country hadn't been called that in 20 years. (In James' defense, his judgment of me was only because I was an idiot American who had called the Czech Republic "Czechoslovakia" numerous times in pre-production email exchanges, even though the country hadn't been called that in 20 years.)

The first hour was terrifying, as big budget sessions can be, because of something every veteran producer knows but I was too tired to emotionally chill about: The orchestra sounded pretty terrible on the first run-through of the first few scenes. This is because recording orchestras generally don't rehearse anything in advance except the ones the music director (James in this case) tells them are particularly challenging. So, on the first run through, you get panic attacks as you think: "Did I just waste $45,000 of my funders' money??? This is my only shot to record this!" With a great orchestra like the one in Prague, however, by the third take they sound amazing. Phew. This realization happens quickly, as you don't record multiple minutes all the way through for such sessions; the orchestra rehearses and records bite-size chunks of a minute here and a minute there, and then we crossfade the best takes together digitally. This could be the subject of an entirely different chapter, but back to the session …

Another strange experience is that you are so used to the sound of the digital version your software has been playing back to you while composing, that the real

Figure 7.14 Back at the Studio

version sometimes sounds weird at first. Good … but *different*. You learn to ignore this quickly, realizing that just because the synth version is familiar doesn't mean it's better. At all. You remind yourself there's a reason you flew across the globe to record with the real thing. Now, when I listen to the software's version compared to the final luminous orchestral recording, it's hilarious that I even had this experience. But our mind loves what's familiar

After the first day of recording and no sleep the night before, I stumbled back to the hotel, did a workout and ate, and crashed for 13 hours of straight sleep. The next day, after more sleep than I'd had in months, was delightful. After two successful full days of recording, the good news is that the orchestra sounded amazing on the recording. The players were stellar, James ran an amazingly tight ship, and the recording engineer and Control Room engineer Jan Holzer in Prague was amazeballs. (Totally a word.) The bad news was that we were several hours behind schedule, making it unlikely that we'd finish recording all the scenes in the half-day session remaining tomorrow; I was out of money and could not hire the orchestra for another half day; and there was the pesky issue that I still had not completed the *final two scenes* completing the storyline and message of the work.

My only solution for this – as with most things – was to simply work with positive energy and do what I could do. We resolved to work faster, and record fewer takes the next day, which would obviously reduce the quality of the performances but give us a shot to record everything. And I pledged to stay up all night – for the second night in three nights – to finish the final scores. Here we go again …

Figure 7.15 In the Session

At 7am before an 8am to noon session, I completed the full score of the last two scenes. But … I had not even started to extract the 30 parts for the different instruments. I was so close, but totally screwed. I sent the Sibelius score files to James and his copyist, told him the status, and grabbed a quick shower in the weird European shower in my hotel room (who builds a shower without a shower door?) and breakfast. Then I stumbled down the street to the orchestra's recording hall, right past the actual theater where Mozart's *Don Giovanni* premiered, an inspirational reminder that I was in a city steeped in music history, yet was probably going to die of this production, much like Mozart died from his *Requiem*, except that mine was an opera for kids about an evil bunny, and Mozart was producing one of the greatest works in history. Whatever. He wrote the overture for *Don Giovanni* the night before the premiere; at least we had that in common.

And this, dear readers, is the moment where the brilliant partnership of the human spirit and modern digital orchestration saves the day. When I arrived, I was told two things: 1. James (who I had thought didn't really like me) had decided that what we were doing – bringing a new operatic work for kids teaching them to love and understand the orchestra – was so good and valuable that he had personally ponied up eight grand to hire the orchestra for another four hours in the afternoon so we could finish the recording at the quality level its music and purpose deserved; and 2. James had brought his copyist into the studio and promised this amazing man would open up the score files of my final two scenes, extract all the parts for the players, and get both the score and parts onto their podiums by the new afternoon session. It was a testament to the previously-unheard-of collaboration that digital orchestrating software allows. If you have the same software and the same knowledge base, you can collaborate magnificently – even though the City of Prague's copyist and I literally didn't even speak the same verbal language.

Figure 7.16 Additional Parts

(Pause for me to cry tears of gratitude to James, his orchestra, his copyist, and the makers of Sibelius as I did that morning when I arrived.… Okay, back to the book …)

That afternoon, after our lunch break, I was more exhausted and grateful than I had ever been. I listened to the final scene produced by 65 musicians, a talented conductor, an accomplished recording engineer, a generous and purposeful music director, and a brain-dead composer. It was the scene I wrote between 3am and 7am that very morning, and despite that, for some strange confluence of reasons, it is perhaps the most beautiful and profound music of the whole two hours and fifteen minutes of the work. It summarizes the purpose of the work brilliantly, and sounds like something beautiful and meaningful that a huge group of people collaboratively pulled off through toil, commitment and trouble. It was amazing, and a testament to the combined power of talent and technology. As Leonard Bernstein famously quipped: "To achieve great things, two things are needed; a plan, and not quite enough time."

Ultimately, my concerns and everyone's commitment – from funders to software programmers to musicians – were rewarded. The recording was extremely well-received – it's the only recording I've ever made that got 100 published reviews, and literally all of them were glowing. It was universally lauded as an old-school recording for kids reminiscent of *Peter & The Wolf* or Britten's *Young Person's Guide To The Orchestra*, *School Library Journal* named it an Essential Recording, it's now a part of the collection of libraries and homes throughout America, and now kids, parents and teachers can take home a copy of an audiophile recording of the work after they see the orchestral shows – or hear it without even going to a show at all. And while I'm proud to feel that its old-school, production-quality street cred is justified – even the idea of a composer writing a full two-act opera for normal newbies who don't drive expensive cars to the opera is so audaciously outdated today as to be a fresh concept – the old-school recording of *Phineas McBoof Crashes the Symphony* would never have materialized without full-scale digital orchestrating and recording technology.

Microphone Characteristics

FREQUENCY RESPONSE

Ideally, microphones should respond equally well to all frequencies across the normal range of human hearing, 20–20,000 Hz. When expressed in a graph that plots output in decibels against frequency, this response appears flat. Although manufacturers find a flat frequency response difficult to attain in large spaces, they can come quite close to it in free-field applications, where direct sound predominates, as shown in Figure 4.1.

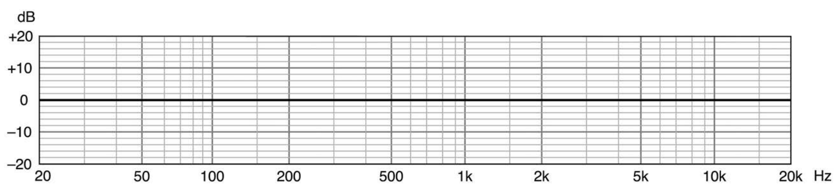

Figure 4.1 Flat frequency response.

However, in the diffuse field, where reverberant sound dominates, rooms tend to absorb treble frequencies, and manufacturers compensate for this phenomenon by enhancing a mic's response to higher frequencies. Microphone designers often boost the sensitivity of omnidirectional mics by 2.0–4.0 dB in the region of 10 kHz (the illustration in Figure 4.2 shows the typical response).

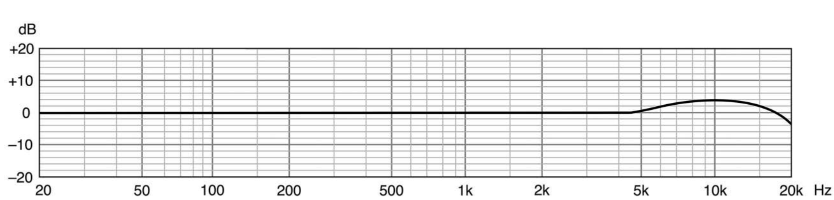

Figure 4.2 Sensitivity boost centered on 10 kHz.

DIRECTIONAL (POLAR) PATTERNS

The polar response of a microphone indicates its sensitivity to sounds arriving from any location around the mic. Manufacturers plot a capsule's response pattern on a polar coordinate graph comprised of 360° concentric circles, each circle representing a difference in sensitivity of a set amount (usually 2.5 or 5.0 dB). They locate the mic's diaphragm at the center and divide the circumference into 30° intervals, which allows them to show the relative sensitivity of the capsule for a given frequency in relation to the angle of incidence (mics respond differently at various frequencies, and the graphs show the amount of attenuation that occurs for specific frequencies in relation to 0° [on-axis]).

Omnidirectional microphones pick up sound equally from all directions, so they have a response pattern quite close to a perfect circle, at least for lower frequencies. The graphs in Figure 4.3 show the typical polar patterns of a small-diaphragm condenser. This mic has a purely omnidirectional pattern below 2.0 kHz, and it exhibits the normal narrowing of response at higher frequencies. This attenuation occurs because soundwaves approaching an omni mic from the rear with wavelengths equal to or less than the microphone's diameter tend not to bend around the end of the mic (see Figure 4.4; for a more detailed discussion, see Ballou 2008: 494–5). Consequently, at 4.0 kHz, the decreased sensitivity of the microphone amounts to just a decibel or so and is virtually unnoticeable, but between 8.0 and 12.5 kHz the drop certainly becomes apparent. Above 16 kHz, the response pattern of the mic approaches unidirectionality.

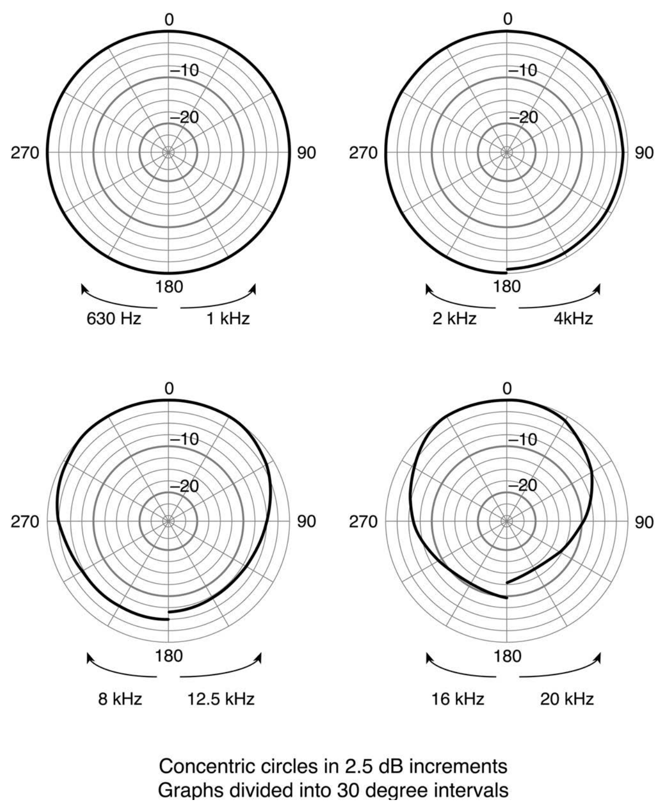

Figure 4.3 Polar patterns at various frequencies for an omnidirectional microphone.

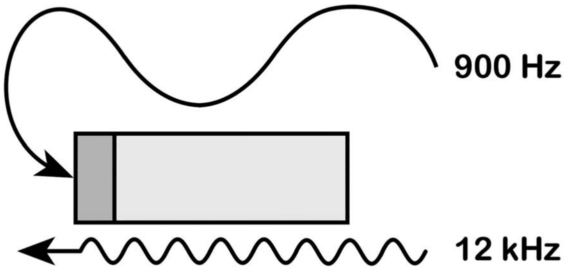

Figure 4.4 Attenuation of soundwaves approaching from the rear of a microphone, when the wavelengths are equal to or less than the microphone's diameter.

However, when recordists wish to restrict the amount of ambient or reverberant sound a mic captures, they generally employ directional models. These microphones can have both primary and secondary areas of pickup, as well as one or two null regions. Cardioid mics feature a reasonably wide pickup area at the front, with a large null at the rear (see the left diagram in Figure 4.5), while supercardioid and hypercardioid microphones restrict the width of their front lobes but have small lobes of limited sensitivity at the rear (the middle diagram in Figure 4.5). Although these latter two capsule types certainly reject sounds from behind, they do so mainly in the regions to the left and right of the rear lobe. Bi-directional mics have two lobes of equal size but opposite polarity, with nulls centering on 90° and 270° (the right diagram in Figure 4.5).

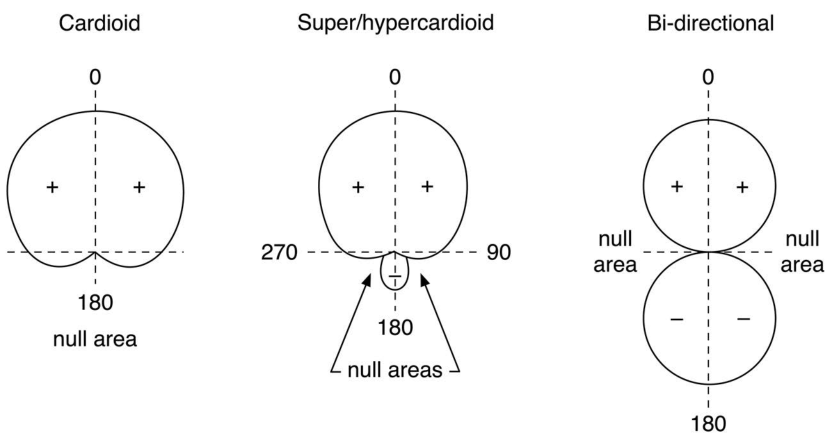

Figure 4.5 Pickup and null areas.

RANDOM ENERGY EFFICIENCY (REE; ALSO CALLED DIRECTIVITY FACTOR)

The efficiency of a mic can be determined by measuring the degree to which the capsule responds to the sound source itself (direct sound) relative to the reverberant field picked up from all directions. Omnidirectional mics respond equally to all sounds, and because of this, manufacturers use them as a reference against which they compare directional mics. Engineers assign omnis an REE of 1, and directional mics, which have a more “selective” response, an REE of less than 1. For example, both bi-directional mics and cardioids react to only about 1/3 of the total sound field, so they receive an REE of 0.333. This means that the ambient sound picked up by these mics is 1/3 or 4.8 dB lower than the direct sound. The narrower supercardioids and hypercardioids have REEs of 0.268 and 0.250 respectively. Supercardioids pick up 5.7 dB less ambient sound, whereas hypercardioids pick up 6.0 dB less.

DISTANCE FACTOR

Compared to an omnidirectional mic, recordists may place a directional transducer farther from a sound source and still produce similar audio results. Directional mics, because of their forward-oriented pickup patterns, reject a great deal of random off-axis sound, and when used in reverberant environments, engineers express the equivalent working distance of a directional microphone, relative to an omni, in terms of its distance factor, an indication of how far recordists can locate a directional mic from a sound source and have it pick up that instrument with the same ratio of direct to reverberant sound as an omni (see Figure 4.6). For example, the distance factor of a cardioid microphone is 1.7, which means that engineers may position it 1.7 times farther away from a sound source than an omni. This greater working distance results from the ability of directional mics to reject off-axis (reverberant) sound.

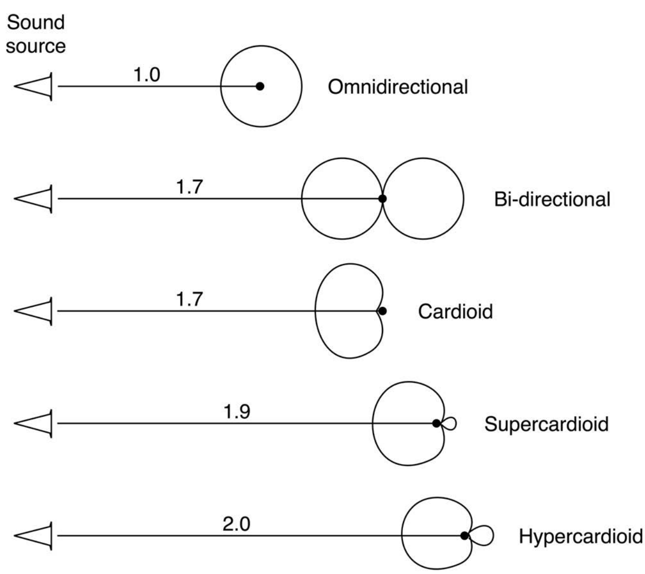

Figure 4.6 Distance factor.

In general, the distance between a microphone and its sound source can dramatically affect the character of a recording. When a recordist positions a mic close to a source, the microphone mainly captures direct sound, a placement that will exaggerate every sound emanating from the source, including imperfections. But if an engineer situates a mic quite a distance from the source, most of the direct sound will be eliminated. In this spot, the microphone "hears" the source as a whole but cannot capture subtle details. Between these locations lies the critical distance, a point at which the level of direct sound equals that of the reverberations (for a fuller discussion of room ambience, see Part 2, Chapter 6).

PROXIMITY EFFECT

All microphones that work on a pressure-gradient principle (bi-directional and cardioid patterns) exhibit proximity effect, a discernible increase in the low-frequency response as a sound source moves closer to the mic. Depending on the design of a microphone's capsule, the effect begins to become audible at distances of around 50 centimeters (20 inches) and is particularly noticeable under about 30 centimeters (12 inches).

A bit of physics, simplified for our purposes, helps to explain the effect. In the direct or free field of an open space (that is, a space free from reflections), soundwaves radiate in all directions from a source in ever-expanding spheres, and as the surface areas of these spheres increase over distance, the intensity of the sound decreases in relation to the area the soundwaves spread across (see Figure 4.7). Physicists define sound intensity as the amount of energy per unit of area (again, measured in terms of sound pressure level), and the Inverse Square Law states that the intensity of a sound decreases proportionally to the square of the distance from the source. In other words, when soundwaves radiate outwardly from a source in the direct field, they spread over a larger area, and for every doubling of the distance, the sound pressure reduces by half, which the human ear perceives as a decrease of 6.0 dB (somewhat less in enclosed spaces). The effect of the Inverse Square Law ends at the point in a room, called the critical distance, where direct sound and reverberation are equal in level (see Part 2, Chapter 6 for a fuller discussion of critical distance).

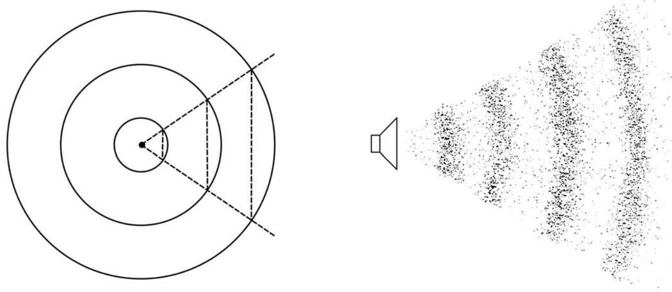

Figure 4.7 Spherical propagation of soundwaves in an open space.

When a recordist positions a microphone close to a sound source, the Inverse Square Law plays an important role in determining the relative pressures soundwaves exert on the front and back of the diaphragm, but when an engineer locates a mic farther from an instrument, the sound pressure on the front and back of the diaphragm approaches, for all practical purposes, equality, simply because the difference in distance that the soundwaves travel to reach either the front or the rear becomes negligible. The Inverse Square Law allows us to understand the principle: a 6.0 dB drop in loudness occurs only when the distance from the source doubles. For example, at 100 centimeters (1 meter), the path from the sound source to the diaphragm is significantly longer than the 1 centimeter path from the front of the diaphragm to the rear (the actual distance from the front to the back varies depending on the design of the capsule). Hence, an increase in the length of travel from 100 to 101 centimeters does not discernibly alter the sound pressure level on the rear side of the diaphragm.

But for shorter distances, the Inverse Square Law plays an important role in determining the relative pressures soundwaves exert on the front and back of the diaphragm, for the front-to-rear dimension is no longer insignificant. John Woram (1989: 102) has determined that a microphone with an internal path difference of 1 centimeter would have discrepancies in sound pressure level between the two sides of the diaphragm of 6.02 dB when placed 10 centimeters from the sound source and 0.83 dB at 100 centimeters. Figure 4.8 illustrates these principles in general terms.

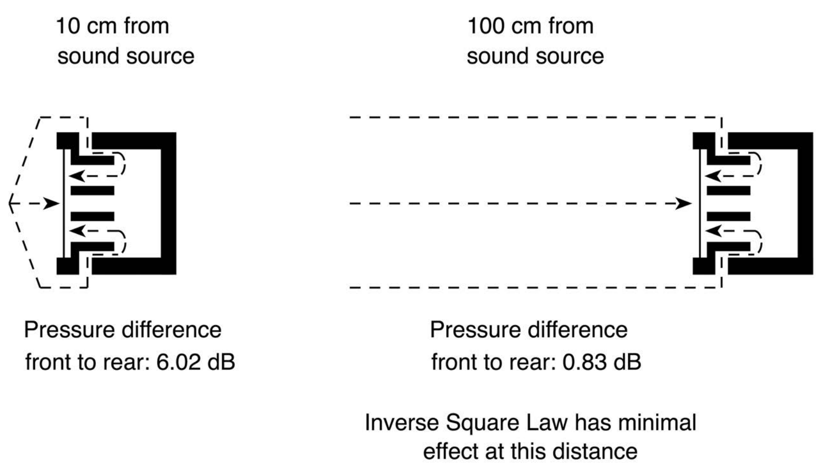

Figure 4.8 Inverse Square Law and discrepancies in sound pressure level.

Note that in Figure 4.8 a close mic placement resulted in 6.02 dB of greater sound pressure at the front of the diaphragm, whereas at 100 centimeters the difference between the front and the back shrank to less than 1.0 dB. Moreover, within the near field, the pressure differences between the front and the back of the diaphragm cause the sound pressure level to rise primarily for lower frequencies. This discrepancy results from phase anomalies across the frequency spectrum of a musical sound. The higher frequencies (shorter wavelengths) of a complex waveform do not bend around the edges of the capsule very easily and reach the two sides of the diaphragm proportionally more out of phase than lower frequencies (soundwaves across the normal range of human hearing vary in length from 1.7 centimeters [0.7 inches] at 20 kHz to 17 meters [56 feet] at 20 Hz). Consequently, a greater amount of cancellation occurs at higher frequencies. Low frequencies, however, consist of waves far longer than the diameter of the diaphragm, and because these waves bend around the capsule quite readily, they arrive at the two sides more in phase. Thus, the lower frequencies tend to sum together, and this summing boosts the low-frequency component of complex wave forms (the next section discusses phase more fully). The graph in Figure 4.9 shows how distance determines the degree of proximity effect below 640 Hz for a typical small-diaphragm condenser.

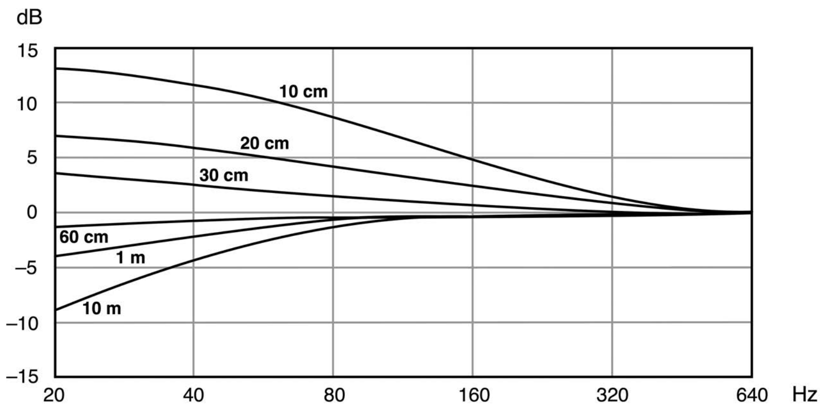

Figure 4.9 Proximity effect in a cardioid microphone.

PHASE

The term phase refers to the starting position of a periodic wave in relation to a complete cycle, and as mentioned above, physicists measure the cyclic nature of waves in degrees (see Figure 4.10).

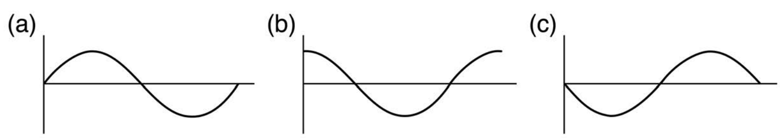

Figure 4.10 Phase in relation to a complete cycle: (a) starting at 0° (b) starting at 90° (c) starting at 180°.

If a soundwave (either simple or complex) arrives at a pair of microphones at the same time, that is, at an identical point in its cycle, the peaks and troughs align perfectly and combine electrically to produce a single waveform of double the amplitude or level (note that when an acoustic wave strikes a diaphragm, the transducer converts it into a similar wave shape within alternating current). Physicists refer to this doubling as constructive interference, and the two signals are "in phase," because they occupy the same relative position in time within a given cycle of the waveform (shown in Figure 4.11 as a single cycle of a sine wave).

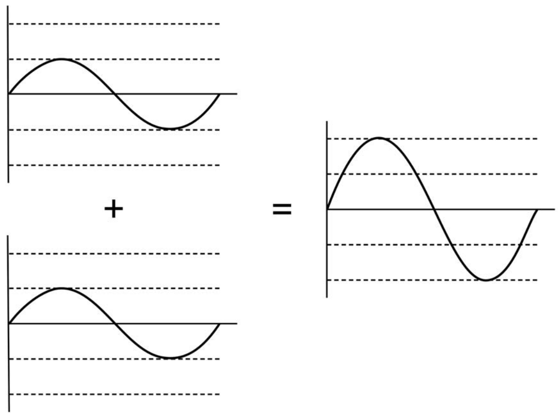

Figure 4.11 Constructive interference, "in phase."

But when a soundwave reaches the two microphones at different times, that is, at different places in its cycle, the peaks and troughs no longer align perfectly. Physicists call this misalignment destructive interference, and it can range from slight to complete. The combination of the two output signals retains the original frequency but at a lower level than an "in phase" configuration. A partial phase reinforcement/cancellation of a sine wave might look like the soundwave shown in Figure 4.12 (compare the lower amplitude in Figure 4.12 with the doubling shown in Figure 4.11).

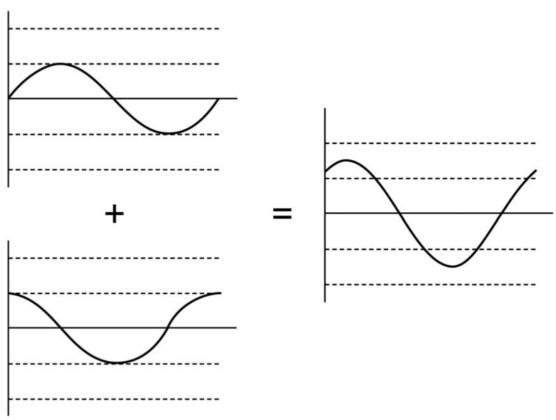

Figure 4.12 Destructive interference, partial reinforcement and cancellation.

Complete phase cancellation occurs when the peaks of one signal coincide exactly with the troughs of the other. Hence, a soundwave reaching one of the mics at the start of its cycle (0°) and the other at 180° produce a combined amplitude of zero (silence), as the two waves are, so to speak, 180° "out of phase" (see Figure 4.13).

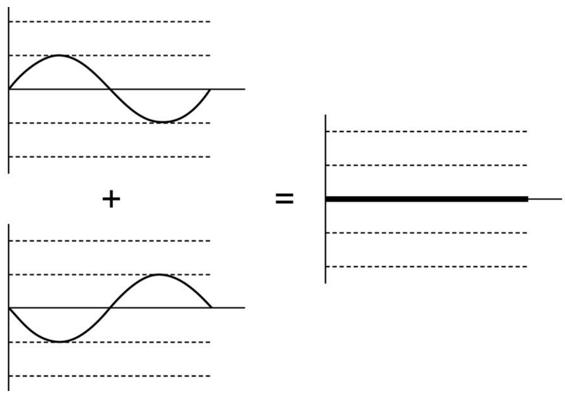

Figure 4.13 Destructive interference, complete cancellation.

The examples shown in Figures 4.10 to 4.13 represent just one frequency from a sound source, but because a spectrum of frequencies creates musical sounds (that is, complex waveforms consisting of multiple sine waves), constructive and destructive interference can occur simultaneously across a range of frequencies. For instance, if the delay in arrival at a second microphone matches the time it takes for one of the frequencies in a complex soundwave to complete a single cycle, a doubling of that frequency's amplitude results (a cycle of 1 ms combined with a delay of 1 ms doubles the amplitude of a 1 kHz frequency, for the wave arrives at the two mics "in phase"). But that same delay cancels the 500 Hz component of the spectrum, because the arrival times are 180° "out of phase." Other frequencies in the complex shift by amounts smaller than 180°, and these intermediate "phase shifts" partially cancel or reinforce the sine waves associated with those frequencies. In fact, short delays occurring in a complex waveform can produce comb filtering, a set of mathematically related (and regularly recurring) cancellations and reinforcements in which the summed wave that results from cutting and boosting frequencies resembles the teeth of a comb. The graph in Figure 4.14 illustrates the comb filtering produced when a signal reaches the second microphone of a pair 1 ms after it arrives at the first mic (peaks of 6.0 dB and dips of 30.0 dB are present).

"Three-to-One" Principle

These phase issues certainly have a considerable impact on the decisions engineers make about the placement of two or more microphones, for if recordists

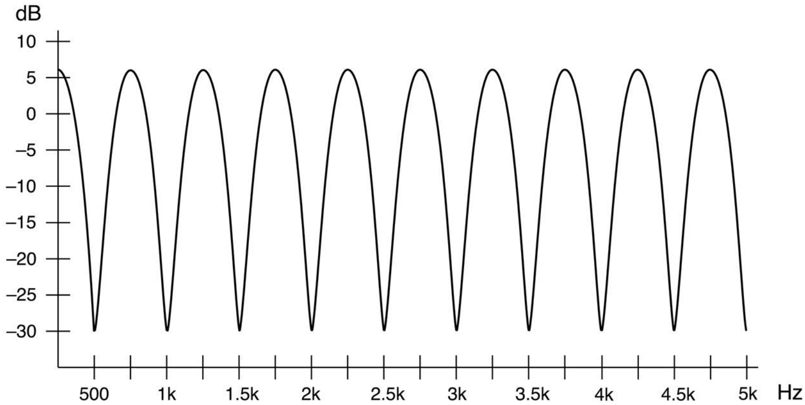

Figure 4.14 Comb filtering.

mistakenly capture out-of-phase signals, various degrees of interference, including comb filtering, can have quite a negative effect on sound quality, especially when phase-shifted signals are either played back or summed to mono. For example, the audibility of comb filtering (that is, the amount of undesirable "phasiness" or "hollowness" induced) depends on the time delay between the two mics: larger delays tend to render comb filtering inaudible, while shorter delays (particularly those under 10 ms) exacerbate the problem. Moreover, when the two mics have identical output levels, boosts of about 6.0 dB and cuts of as much as 30.0 dB frequently occur (as shown in Figure 4.14).

Engineers can, of course, minimize the negative effect of this distortion by reducing the difference in amplitude between the peaks and troughs caused by phase shifts. Phasiness becomes less prominent when the output level of one transducer is lower than the other, and an attenuation of 9.0 to 10.0 dB at one of the mics can decrease amplitude differences to about 4.0 dB, a level at which the worst offender, comb filtering, becomes tolerable for most listeners (some say it resembles a pleasant room ambience). In fact, by placing two microphones in accordance with the "three-to-one" principle, recordists can easily achieve 9.0 dB of attenuation.

Figure 4.15 demonstrates how audio researchers arrived at this principle through tests carried out in an anechoic chamber (the information in the following paragraphs has been taken from Burroughs 1974: 115–19). The technicians placed a sound source directly on-axis 2 feet (61 centimeters) in front of a microphone (number 1 in Figure 4.15) and recorded the sound through that mic. They then recorded the source five more times with two microphones, the second one located 2 feet (61 centimeters), 4 feet (122 centimeters), 6 feet (183 centimeters), 8 feet (244 centimeters), and 10 feet (305 centimeters) away from the on-axis mic.

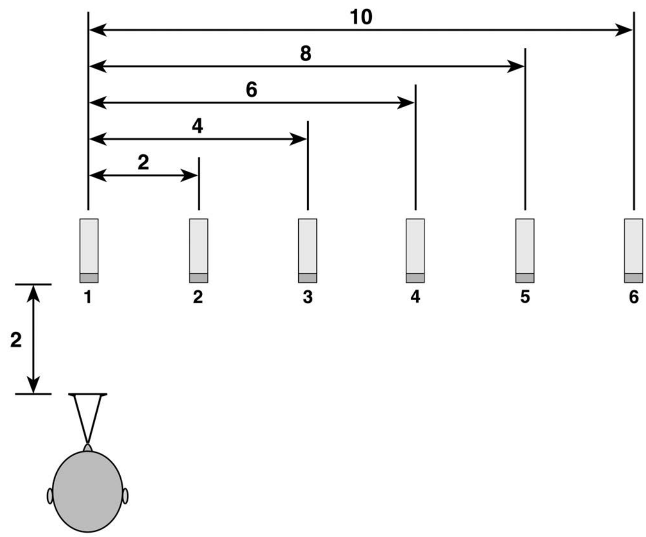

Figure 4.15 Testing procedure for the "three-to-one" principle.
Source: Adapted from Burroughs 1974: 117.

When plotted on a graph (see Figure 4.16), the on-axis response of microphone number 1 in Figure 4.15 produced the curve shown at distance "0" (the top contour). The blended signals from the two mics generated the frequency-response curves given below it (the numbers at the ends of the contours indicate the distance between the microphones).

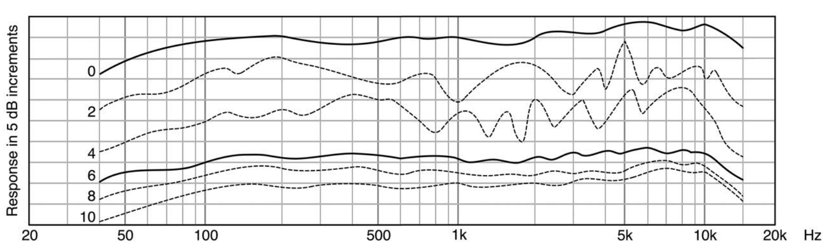

Figure 4.16 Frequency response curves.
Source: Adapted from Burroughs 1974: 118.

These curves illustrate the varying amounts of deterioration that combined signals can cause. When the researchers located the second transducer either 2 feet (61 centimeters) or 4 feet (122 centimeters) from the on-axis mic, noticeable reinforcements and cancellations occurred, and only when they placed the second microphone 10 feet (305 centimeters) away did the combined signals parallel the on-axis response of mic number one. However, the response curves at 6 feet (183 centimeters), 8 feet (244 centimeters), and 10 feet (305 centimeters) resemble one another, and despite the subtle differences in detail that the scientists measured in the anechoic chamber, skilled listeners did not notice audible improvements in sound quality when the researchers positioned the second microphone farther than 6 feet (183 centimeters) away. At 6 feet (183 centimeters), the distance between the two microphones is three times as great as the 2-foot (61-centimeter) distance from the on-axis mic to the sound source. Thus, the test revealed that the engineers did not need to have more than a three-to-one ratio between mics to avoid noticeable phase interference.

In a multiple microphone setup, recordists can establish this ratio with the distances (all in feet) indicated in the diagram shown in Figure 4.17 (in practice, one may need to modify the placement somewhat, for the theoretical model does not take into account differences in loudness between the sound sources).

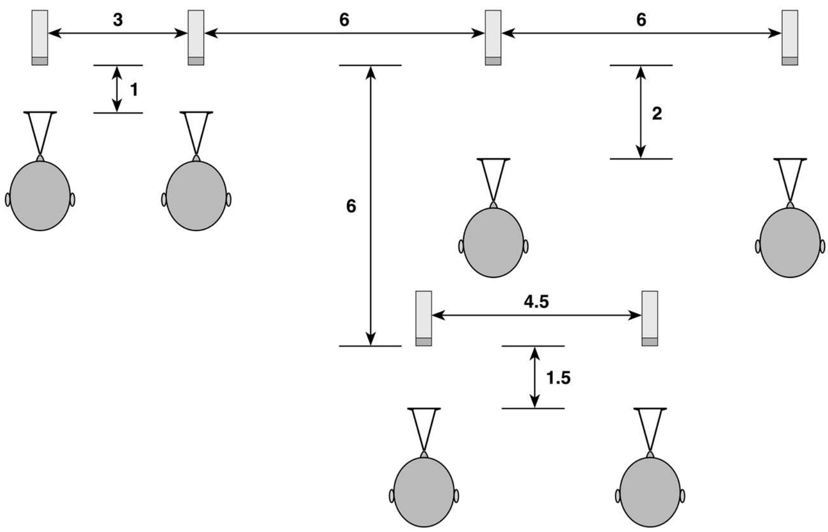

Figure 4.17 Multiple microphone setup to avoid comb filtering.
Source: Adapted from Burroughs 1974: 118.

These locations should result in a difference of 9.0 to 10.0 dB between microphones; that is, the background sounds picked up by any given mic should be at least 9.0 dB lower than the primary sound source in front of the microphone. During rehearsal, recordists can check this attenuation one mic at a time. With every microphone switched off except for one, the level at the sole remaining mic should drop by at least 9.0 dB when the performer at that microphone stops playing (in other words, the mic should "hear" the background sounds at a level 9.0 dB lower than the sound produced by the player directly in front of the microphone).

In summary, if recordists wish to maintain sufficient phase integrity in concert/recital hall locations, they simply need to follow the "three-to-one" principle: for every unit of distance between a microphone and its source, separate nearby mics by at least three times that distance. However, in situations where engineers opt for close miking and/or further processing that requires signal splitting, recordists regularly correct small amounts of phase shift during post-production either by manually moving the signals to synchronize them or by employing a plugin, such as Sound Radix's *Auto Align*, to bring the signals into an acceptable phase relationship.

Auto Align analyzes pairs of signals to find the amount of time delay between them and then compensates for the difference (see the screenshots in Figure 4.18). The plugin presents its measurement in samples, milliseconds, and distance (centimeters and inches), and beyond correcting out-of-phase signals, it allows engineers to "enhance the sense of space" when some form of delay is desirable. In other words, *Auto Align* can "time-place the [distant] microphones to better match the close-mic'd source" (Sound Radix n.d.: 2).

Figure 4.18 *Auto Align*, two signals compared and phase-aligned (72 sample delay between them).

Source: Used with the permission of Sound Radix.

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Classical Production

Situating Historically Informed Performances in Small Rooms

Robert Toft

One of the challenges facing performers who wish to record works from the past in an historically informed manner centers on finding acoustic spaces similar to those in which the music would have been heard originally. Unfortunately, many musicians today regard large churches as ideal locations for recording much of the music composed in the Renaissance and Baroque eras, even though these rooms can be far too reverberant for a significant portion of the repertoire, especially solo songs accompanied by quiet instruments (such as the lute, guitar, or harpsichord). Curiously, artists often spend a great deal of time learning how to bridge the gap between the written descriptions of performing practices found in old treatises and actual performance yet choose less-than-optimal locations to record vocal music that seems to have been conceived for intimate spaces (for drawings of the modest rooms in which Giulio Caccini probably sang his own compositions in the late 16th and early 17th centuries, see Markham 2012: 200–203).

In fact, recent research has shown that musicians in the 16th to 18th centuries regularly performed in private chambers or small music rooms (Howard and Moretti 2012: 106–107, 111–114, 185–189, 200–203, 248–249, 320), and because many of the *camere per musica* measured no more than seven by eleven meters (23 x 36 feet), with a ceiling height of five to six meters (16 to 20 feet), their volume (approximately 385 cubic meters or 13,250 cubic feet) produced a reverberation time of less than a second, which means that these spaces tended more to clarity and intimacy than reverberance (Orlowski 2012: 157–158). Moreover, since listeners would have been seated close to the performers in these rooms, direct sound would predominate, and early reflections from the walls and floor would further contribute to the sonic impression of clarity and intimacy (Orlowski 2012: 158).

Most recordists, however, do not have access to intimate historic spaces for recording purposes, but they can employ modern studio technology to replicate the aural sense of these *camere*, not only through convolution reverbs based on impulse responses taken from rooms in 16th- to 18th-century buildings but also with artificial reverberation of the reflection-simulation type designed to mimic the characteristics of generic spaces. This chapter discusses the procedures a group of recordists followed, first to create an historically informed interpretation of a 17th-century solo song and then to place that performance in an acoustic that would make listeners feel as though they were sitting in the same small room as the performers. Specifically, the chapter focuses on a track from Studio Rhetorica's recording *Secret Fires of Love*, which features tenor Daniel Thomson and Baroque guitarist Terry McKenna under my musical direction (Talbot Productions, TP 1701, 2017). The recording of "Si dolce è'l tormento", a song composed by Claudio Monteverdi and published in Carlo Milanuzzi's *Quarto scherzo delle ariose vaghezze* (Venice, 1624), was produced by me and recorded and mixed by Robert Nation (Kyle Ashbourne, assistant engineer) at EMAC Recording Studios in London, Canada (the track is available on my website, www.talbotrecords.net, or in full-track preview on CD Baby). I am most grateful to Robert for generously sharing his philosophies/procedures of recording and mixing with me, as the production and post-production sections of this chapter could not have been written without his input, for they blend Robert's explanations of his practices with my contributions as producer.

The first part of the chapter considers the pre-production phase of the project, during which Daniel, Terry, and I finalized the interpretive strategies that would be employed, and the following sections focus on the ways studio production practices at EMAC helped us transfer our historically informed conception of "Si dolce è'l tormento" to disk.

PRE-PRODUCTION

Older principles of interpretation differ considerably from those currently used by classical musicians, and in order for people interested in historical performance to recover the old methods, they must reconstruct the practices from surviving sources of information. Fortunately, a great deal of material comes down to us, and this has allowed us not only to root the interpretive strategies employed in *Secret Fires of Love* in period documents but also to take a fresh approach to Renaissance and Baroque songs. Recent research has shown that from the 16th to the early 19th centuries, singers modeled their art directly on oration and treated the texts before them freely to transform inexpressive notation into passionate musical declamation (see in particular Toft 2013, 2014).

Daniel adopts the persona of a storyteller, and like singers of the past, he uses techniques of rhetorical delivery to recreate the natural style of performance listeners from the era probably would have heard. This requires him to alter the written scores substantially, and his dramatic singing combines rhetoric and music in ways that sympathetically resonate with performance traditions from the Renaissance and Baroque eras. In "Si dolce è'l tormento", Daniel sings prosodically, emphasizing important words and giving the appropriate weight to accented and unaccented syllables;

employs a highly articulated manner of phrasing; alters tempo frequently through rhythmic *rubato* and the quickening and slowing of the overall time; restores *messa di voce*, the swelling and diminishing of individual notes, as well as phrases, to its rightful place as the "soul of music" (Corri 1810: i. 14); contrasts the tonal qualities of chest and head voice as part of his expression; and applies *portamento*.

Among these principles, highly articulated phrasing, alterations of tempo, and variations in the tonal quality of the voice represent the most noticeable departures from modern practice. Singers of the past inserted grammatical and rhetorical pauses to compartmentalize thoughts and emotions into easily discernible units (that is, stops at punctuation marks and in places where the sense of the sentence called for them), and this frequent pausing gave listeners time to reflect on what they had just heard so they could readily grasp the changing sentence structure. In 1587, Francis Clement explained the rationale behind the addition of unnotated pauses: "the breath is relieved, the meaning conceived . . . the eare delited, and all the senses satisfied" (pp. 24–25; for further information on pausing, see Toft 2013: 20–45; Toft 2014: 84–98). Moreover, writers from Nicola Vicentino (1555: fol. 94v) to Giambattista Mancini (1774: 150) observe that singers best convey the true sense and meaning of words in a natural way if they derive the pacing of their delivery from the emotions in each text segment. Or to use Vicentino's words, tempo fluidity has "a great effect on the soul/effetto assai nell'animo" (1555: fol. 94v).

Similarly, the use of appropriate vocal timbres to carry the text's emotions to the ears of listeners requires singers not only to differentiate their registers (so that the lowest and highest parts of the range contrast with the middle portion) but also to link timbre and emotion (smooth and sweet, thin and choked, harsh and rough) – "the greater the passion is, the less musical will be the voice that expresses it" (Anfossi c.1840: 69). In earlier eras, a versatile tonal palette prevented the monotony of what David Ffrangcon-Davies dismissed in 1905 as the "school of sensuously pretty voice-production". Indeed, as Ffrangcon-Davies suggests, the then new monochromatic approach to timbre meant that if audiences had heard a singer in one role, they had heard that singer in every role (pp. 14–16).

Armed with a collection of historic principles, our first task in recreating period style was to study the lyrics of "Si dolce è'l tormento" to find all the places a singer might wish to insert grammatical or rhetorical pauses (see Figure 7.1).

This exercise involved following principles described in treatises on rhetoric and oration to add pauses of varying lengths at points of punctuation (grammatical pauses) and to separate subjects from verbs and verbs from objects, as well as conjunctions, relative pronouns, participles, and prepositional phrases from the text that precedes them (rhetorical pauses). The compartmentalization of ideas and emotions organizes and paces the content of the poem so that listeners can easily grasp the story, and since some ideas require a slower or quicker delivery than others, compartmentalization also provides appropriate places for singers to change the speed of delivery to match the emotional character of the phrases.

1.	Si dolce * è'l tormento *	So sweet * is the torment *
	ch'in seno mi stà *	which resides in my breast *
	ch'io vivo contento *	that I live in contentment *
	per cruda beltà. *	because of the cruel beauty. *
	Nel ciel di bellezza *	In the heaven of beauty, *
	s'accreschi fierezza *	if arrogance increases *
	et manchi pietà *	and pity decreases, *
	che sempre qual scoglio *	then always like a rock *
	all'onda d'Orgoglio *	in a wave of disdain *
	mia fede * sarà. *	my faith * shall exist. *
2.	La speme fallace *	False hope *
	rivolgam'il piè *	overwhelms my foundation, *
	diletto, * ne pace	neither delight * nor peace
	non scendano a me *	comes to me, *
	e l'empia * ch'adoro *	and the wicked one * that I adore *
	mi nieghi ristoro *	denies me the comfort *
	di buona mercé: *	of kind mercy: *
	tra doglia infinita *	amidst infinite pain, *
	tra speme tradita *	amidst hope betrayed *
	vivra * la mia fe. *	my faith * shall have life. *
3.	Per foco, * e per gelo *	Because of the fire, * and because of the ice *
	riposo non hò *	I have no rest *
	nel porto del Cielo *	[but] in the refuge of heaven *
	riposo haverò. *	I shall have rest. *
	Se colpo mortale *	If a deadly blow *
	con rigido strale *	with a sturdy arrow *
	il cor m'impiagò, *	wounds my heart, *
	cangiando mia sorte *	changing my fate *
	col dardo di morte *	with the dart of death *
	il cor * sanerò. *	shall heal * my heart. *
4.	Se fiamma d'Amore *	If the flame of love *
	già mai non * sentì. *	never before * was felt. *
	Quel riggido core *	That merciless heart *
	ch'il cor mi rapì. *	who stole my heart. *
	Se nega pietate *	If you deny pity, *
	la cruda beltate *	the cruel beauty *
	che l'alma invaghì *	who ravished my soul, *
	ben fia * che dolente *	it shall be right * that in sorrow, *
	pentita, * e languente *	repentant * and languishing, *
	sospirimi * un dì. *	you will sigh for me * one day. *

Figure 7.1 Text of "Si dolce è'l tormento" with grammatical and rhetorical pauses marked; asterisks represent pauses of various size

But apart from organizing and pacing ideas and emotions, singers must decide which word or words within a phrase should be emphasized, and treatises usually combine the discussion of emphasis with that of accent. Accent denotes the stress placed on a single syllable to distinguish it from the others in a word (this is known as speaking or singing prosodically), whereas emphasis refers to the force of voice laid on an entire word or group of words to bring the associated ideas to the attention of listeners. Proper accentuation, then, adhered to the normal pronunciation of words in ordinary speech, while emphatic delivery varied according to the meaning performers wished to convey (for more information on accent and emphasis, see Toft 2013: 73–79, 2014: 98–108).

Emphatic words, then, receive the greatest force within a sentence, and these important words are situated in an overall hierarchy of emphasis in which speakers reserve the strongest sound of voice for the most significant word or idea, firmly and distinctly pronouncing substantives (nouns), adjectives, and verbs, while relegating unimportant words (the, a, to, on, in, of, by, from, for, and, but, and so on) to relative obscurity (Walker 1781: ii. 15, 25). This hierarchy allows performers not only to arrange words into their proper class of importance but also to achieve a distribution of emphases that would prevent sentences from being delivered monotonously with uniform energy (Herries 1773: 218). Thus, the application of accent and emphasis creates light and shade and helps speakers (and singers) clearly project the meaning of long and complex ideas. Inappropriate shading would force listeners to decipher a sentence's meaning from an ambiguous or confusing delivery (Murray 1795: 153).

After we completed our analysis of the song's text using the principles just discussed, Daniel proceeded to create a dramatic spoken reading of the poem. Initially, this meant deciding which pauses would be employed and what ideas would be exhibited prominently (that is, emphasized), as well as what variations in the speed of delivery would suit the changing emotions of the text. As part of this process, Daniel also took note of where *messa di voce* and *portamento* occurred as he spoke, for in these places both the swelling and diminishing of the voice and the sliding between pitches would sound the most natural in singing (*messa di voce* and *portamento* are discussed more fully in Toft 2013: 45–69). Once we were satisfied with the spoken narrative, we transferred Daniel's interpretation to the song, altering Monteverdi's melodic lines to accommodate the dramatic reading.

By rooting our performance in historical documents, we were able to model our understanding of the relationship between performer and score directly on principles from the past. Indeed, singers in earlier times viewed scores quite differently from their modern counterparts. They realized that because composers wrote out their songs skeletally, performers could not read the notation literally, and to transform inexpressively written compositions into passionate declamation, vocalists treated texts freely and personalized songs through both minor and major modifications. In other words, singers saw their role more as one of recreation than of simple interpretation, and since the final shaping of the music was their responsibility, the songs listeners heard often differed substantially from what appeared in print (Toft 2013: 4–6 discusses the relationship between notation and performance).

Composers of the past did not notate subtleties of rhythm, phrasing, dynamics, pauses, accents, emphases, tempo changes, or ornamentation. Clearly, they had no desire (or need) to capture on paper the elements of performance that moved listeners in the ways writers from the time described. In the middle of the 16th century, Nicola Vicentino commented that "sometimes [singers] use a certain method of proceeding in compositions that cannot be written down"/"qualche volta si usa un certo ordine di procedere, nelle compositioni, che non si può scrivere" (1555: fol. 94v), and along these lines, Andreas Ornithoparchus, writing in 1517, praised singers in the Church of Prague for making "the Notes sometimes longer, sometime[s] shorter, than they should" (p. 89 in John Dowland's translation). Around 1781, Domenico Corri characterized the relationship between performance and notation candidly: "either an air, or recitative, sung exactly as it is commonly noted, would be a very inexpressive, nay, a very uncouth performance" (vol. 1, p. 2). Charles Avison had already made this notion explicit in 1753 (p. 124): "the Composer will always be subject to a Necessity of leaving great Latitude to the Performer; who, nevertheless, may be greatly assisted therein, by his Perception of the Powers of Expression", and a hundred years later, voice teachers like Manuel García (1857: 56) continued to suggest the same thing – performers should alter pieces to enhance their effect or make them suitable to the power and character of an individual singer's vocal capability.

In 1555, Nicola Vicentino suggested why performers valued flexibility of tempo (fol. 94v):

> The experience of the orator teaches this [the value of changing tempo (mutare misura) within a song], for one sees how he proceeds in an oration – for now he speaks loudly and now softly, and more slowly and more quickly, and with this greatly moves his auditors; and this way of changing the tempo has a great effect on the soul./La esperienza, dell'Oratore l'insegna, che si vede il modo che tiene nell'Oratione, che hora dice forte, & hora piano, & più tardo, & più presto, e con questo muove assai gl'oditori, & questo modo di muovere la misura, fà effetto assai nell'animo.

Hence, vocalists sang *piano e forte* and *presto e tardo* not only to conform to the ideas of the composer but also to impress on listeners the emotions of the words and harmony, and Vincenzo Giustiniani (c.1628: 108) characterized the approach singers from Mantua took in the latter part of the 16th century:

> [B]y moderating and increasing their voices, forte or piano, diminishing or swelling, according to what suited the piece, now with dragging, now stopping, accompanied by a gentle broken sigh, now continuing with long passages, well joined or separated [that

is, legato or detached], now groups, now leaps, now with long trills, now with short, and again with sweet running passages sung softly, to which one unexpectedly heard an echo answer/col moderare e crescere la voce forte o piano, assottigliandola o ingrossandola, che secondo che veniva a' tagli, ora con strascinarla, ora smezzarla,

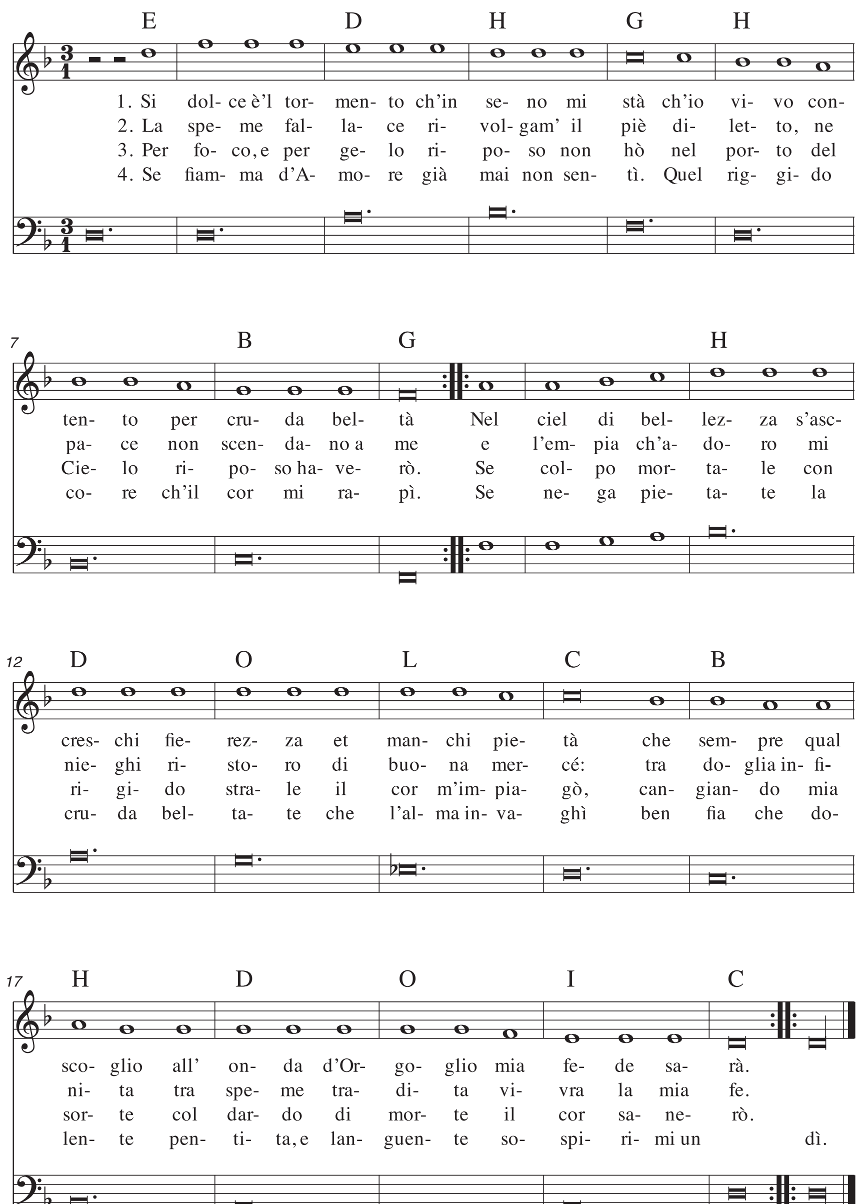

Figure 7.2 Claudio Monteverdi, "Si dolce è'l tormento" (Milanuzzi 1624); letters above each system indicate the chords the guitarist should play.

> con l'accompagnamento d'un soave interrotto sospiro, ora tirando passaggi lunghi, seguiti bene, spiccati, ora grupi, ora a salti, ora con trilli lunghi, ora con breve, et or con passaggi soavi e cantati piano, dalli quali tal volta all'improvviso si sentiva echi rispondere.

Primarily, the period-specific alterations we made to Monteverdi's text and melodic lines involved adding pauses and adjusting the rhythmic values of the notes so that the delivery of the syllables and words came as close to speaking as possible. But Daniel also varied tempo along the lines Vicentino and Giustiniani had suggested and employed light and shade (accent and emphasis) in an historic way, for if he were to sing the melodies exactly as Monteverdi had notated them, he would, in Domenico Corri's 18th-century view, be guilty of an "inexpressive" and "uncouth" performance (a modern edition of Monteverdi's skeletal notation for "Si dolce è'l tormento" appears in Figure 7.2). By placing his persona as a storyteller in this older guise, Daniel has provided what one writer, John Addison (c.1850: 29), called the "finish" to the song in a way that approximates early 17th-century style.

PRODUCTION

The main goal in producing "Si dolce è'l tormento", as well as the other songs included in *Secret Fires of Love*, was to enhance period interpretation through modern studio practices, especially isolated sound sources recorded by closely placed microphones. We chose to blend the worlds of recording and "live" performance so that we could capture a dramatic reading of the song, while achieving sonic clarity. In other words, we did not consider the two activities to be mutually exclusive, and because we felt that making records and archiving a "live" event differed fundamentally, we decided to use punch-ins to perfect excellent takes rather than completely re-record those sections that contained minor imperfections.

From these perspectives, the project benefited from having one person assume the roles of music director and producer, for a single conception of the song could then emerge from the various sonic possibilities available to the artists and engineers. Indeed, decisions made throughout the process, from those that led to an historically relevant interpretation of the printed score to those that guided the design of the soundscape in which the performance was presented, came from imagining how one world might inform the other. Knowledge of both historical performance and recording practices focused the energies of everyone involved in the project on an idealized conception, and the various elements of production, when combined with sympathetic strategies for editing and mixing, helped shape the recording along historical lines.

Robert tracked in ProTools HD at 24 bits / 96 kHz to provide an excellent signal-to-noise ratio, as well as an increased dynamic range, and microphone selection and placement figured prominently at the beginning of the sessions. Since Baroque guitars can be somewhat “noisy” to record, the question arose of how we might best capture the sound of the instrument. The mics chosen would need to produce as “natural” a stereo sound as possible when positioned closely, so microphones with too much boost on the top end would not be suitable. In fact, a stereo pair of omnidirectional microphones with a linear frequency response would probably be ideal for this application, as omnis would allow the characteristic resonance of the instrument to be portrayed realistically, without proximity effect. For the voice, a microphone that could provide a consistent frequency response across Daniel’s range, while keeping sibilance to a minimum, would be preferable, and both ribbon and omnidirectional microphones were obvious possibilities.

A “shoot-out” using mics from EMAC’s and the author’s collections resulted in the following choices:

Voice – Royer R-122 active ribbon mic (a Schoeps small-diaphragm condenser with an MK 2 capsule, omnidirectional with a flat frequency response, was also quite attractive)
Baroque guitar – stereo pairs of DPA’s omnidirectional 4006A condenser and Royer’s R-122 active ribbon

Daniel sang in an isolation booth with his microphone, shielded by a pop screen, placed approximately 12 inches (30 cm) in front of his mouth. Although ribbon mics exhibit a fairly strong proximity effect, the Royer R-122 not only provided the consistency of frequency response we desired but also eliminated most of the problems associated with sibilance. Terry performed in the main tracking room, and because the dynamic range of Baroque guitars is not large, the first pair of mics for his guitar (DPA 4006A) were placed next to the instrument in an A-B arrangement, one pointing at the lower bout and the other at the upper bout. A second pair of stereo mics, R-122s in Blumlein configuration, were positioned about 24 inches (61 cm) from the guitar along its center line to create the sense of space a two millisecond delay would produce.

Once appropriate levels had been set, the tracking procedure consisted of several initial takes of the whole song. The entire team then listened to these recordings to determine if one of them could become a master take that would be refined through punch-ins. At this point, Terry suggested he try some new accompaniment ideas that had occurred to him as he listened (his part was improvised from the chord symbols shown in the preceding score), and since the take that resulted from this extemporaneous performance, complete with an impromptu introduction, had the level of spontaneity we were seeking and did not need any punch-ins, it became the master track which would be edited and mixed.

POST-PRODUCTION

The editing process not only helped us achieve our ideal historically informed conception of “Si dolce è’l tormento” but also allowed us to reduce any noise in the recording that might distract listeners. Because audio recordings lack the visual connection of "live" performance, noise that mars the sensory surface of a disk can be much more disruptive, especially string noise and extraneous breathing in guitar tracks or plosives in vocal tracks. Kyle Ashbourne, the assistant engineer, and I carefully listened to the master take and used the spectral repair feature of iZotope's *Rx Advanced* to reduce string noise to an amount a listener would hear when seated 10–12 feet (three to four meters) away from the performers. In other words, we wished to present the guitar from the perspective of the listener rather than the player.

Similarly, plosives that resulted from the singer's closely placed ribbon microphone were reduced using the high-pass filter in Universal Audio's *Massenburg DesignWorks MDWEQ5*. Moreover, Pro Tools' clip gain helped us achieve the prosodic manner of vocal delivery prized in the past, for we lowered the level of those syllables that close micing had heightened. In addition, whenever the Royer R-122 exaggerated a *messa di voce* with too much energy at its peak, automation of the dynamics in ProTools helped bring those phrases into line with the way they would be heard a short distance from a singer (here we were thinking of how the inverse square law affects the propagation of the sound waves to soften the swells naturally).

The mixing sessions focused on three main elements – reverb, compression, and EQ. As we began to consider possible models for creating a suitable ambience, we decided to listen to a number of lute and Baroque guitar performances that had been recorded in large churches, the typical locations for such recordings. These rooms were, of course, far too reverberant for our purposes, and we realized that without appropriate models to emulate, we needed to imagine a performance space that did not yet exist on a recording. Robert then set about designing an artificial ambience that would approximate the small rooms in which the music was often performed originally.

He chose the same two algorithmic reverbs for the voice and the Baroque guitar, so that Daniel and Terry would sound like they were performing together in a room. The first was the Large Hall B in Universal Audio's *Lexicon 224 Digital Reverb*, with a decay time of 2.0 s in the vocal and 1.7 s in the guitar, and the second was the Small Plate 2 in Eventide's *UltraReverb*, set to a short decay time of 837 ms to "tighten" the ambience surrounding both performers. Robert used the two reverbs quite subtly, mixing them in at a low level, and because Daniel has a large dynamic range, Robert compressed the input to the *Lexicon 224* to bring the level of the loudest passages down by 2–3 dB, so that Daniel's voice would not over-trigger the reverb. Compression, then, helped make the vocal ambience less obvious. Baroque guitars, in contrast, have quite a small dynamic range, and since they could never over-trigger the *Lexicon 224*, Terry's input did not need to be compressed.

Beyond the reflection simulation Robert crafted to place the performers in an appropriately sized room, he included two parallel effects busses on the vocal track and adjusted the frequency balance of Daniel's voice. A compression back bus, combining Universal Audio's EL7 FATSO Sr with Sonnox's Oxford Inflator, was mixed in at a very low level (-24.0 dB) to increase the intimacy of the quietest passages, and on another bus, Nugen Audio's Stereoizer created a bit of extra space in the mix. Specifically, delays of 12 ms on one side and 19 ms on the other added the sense of early reflections, which, together with the reverbs, compensated for recording into a baffle.

To lessen the proximity effect inherent in the Royer R-122, the Massenburg DesignWorks MDWEQ5 parametrically removed some of the energy around 310 Hz, while the high-pass filter of the same plugin disposed of any rumble below 40 Hz. Close micing of a singer can also produce some mild harshness at the loudest moments, and a cut of 1.4 dB in Brainworx's dynamic equalizer dynEQ V2 was used to alleviate this tension around 2357 Hz. Brainworx (n.d.: 4) describes a dynamic EQ as "a filter that is not limited to being set to a specific gain level, but which changes its gain settings dynamically – following the dynamics of a certain trigger signal". Robert also applied the dynEQ V2 plugin to the guitar track to lessen the effect of some low "thumps" around 119 Hz (a cut of 2.5 dB).

On the master bus, because downstream codecs can increase the peak level of the signal somewhat, a true-peak limiter (Nugen Audio's ISL 2) was set at -0.7 to leave room for file conversion, and the general loudness characteristics of the track were analyzed through Nugen Audio's MasterCheck Pro.

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In our recording of "Si dolce è'l tormento", as well as in the other tracks on *Secret Fires of Love*, we clearly embraced the notion that "the most important reverberation exists within the recording, not the playback space" (Case 2007: 263), and since we did not want listeners to experience the music from a distance, as if they were in a large church or concert hall, we decided to create an artificial ambience that would situate them about 10 to 12 feet (three to four meters) from the artists. Hence, a blend of close micing and the digital processes described earlier, all in the service of an historically informed performance, allowed everyone involved in the project (artists, producer, and engineers) to realize on disk what we imagined someone in the 17th century might have heard in the small rooms in which the music was frequently performed.

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The Role of the Producer in Classical Music Recording

Very few people outside of classical music recording actually understand the function of the producer through the preparation, the recording session, and the post production stages of editing and mixing. For smaller-scale projects and recordings with limited budgets, it is becoming more and more common for one person to manage the roles of both producer and engineer. In the world of popular music recording, the producer's job description is generally very different. In this chapter, the classical producer's role is fully explained, and general advice is offered for those engineers and musicians who wish to develop a career in classical music production. The relationships of engineer and producer, and producer and artist, are discussed in detail.

First and foremost, the producer is normally responsible for overseeing all stages of the recording process, and delivering a finished master of the final product to the client (either a record label or the artist themselves). In almost all cases the producer is the lead member of the recording team. While the engineer might commonly be hired or requested by the producer, a label or artist might approach the team members separately in terms of availability and budget concerns. In either case, the engineer should be ready to consult the producer on most decisions involving scheduling and rental costs, as well as any major technical considerations, because in the end the producer is ultimately accountable.

13.1 Qualifications of a Classical Music Producer

A well-qualified candidate for the field of classical music production is someone with a formal education in music and many years of performance experience. A musicologist may know the repertoire, but if they lack playing experience they will be unable to closely relate to the artist's life on stage, the hours in the practice room, as well as many of the physical considerations of a busy performer.

Understanding recorded sound is the second most important requirement. Some years of experience with critical listening and analysis is crucial when it comes to forming an opinion of a musical performance captured by microphones and reproduced over loudspeakers or headphones. This important when it comes to evaluating the overall presentation of an audio recording, and for monitoring consistency in the sound throughout the process of a recording project.

The process of record production must be fully understood as well, specifically the advantages and limitations of the editing and mixing phases, and how this may affect decisions made during the recording session. A very simple example is that when editing between two versions or "takes" of a particular passage, the tempos will need to be the same—otherwise a noticeable shift will occur at the desired edit point in the music. The producer can reduce this artifact by carefully monitoring the tempo of each take during the recording session. Certain irregularities in balance might be adjusted during the mixing, while drastic differences in dynamics between takes might cause problems in the editing and mixing—the producer will need to be aware of this and take appropriate notes during the recording.

The producer of a classical music recording may also be tasked with creating a financial budget including the cost of the hall, the musicians, the engineer, equipment, travel costs, and an estimate of time needed for post production (editing, mixing). These additional tasks are being asked of the producer more often than in the past, as record labels are shrinking and more artists are financing and running their own recording careers. In certain cases the producer may be involved in preparing contracts for the musicians, but only where the appropriate expertise exists.

Almost all professional orchestras are unionized, and as such there are strict rules governing recording session proceedings. Even non-union ensembles will have a preferred structure as to how they record, and the producer needs to be informed. In North America there are strict rules in place for recording orchestras that are unionized members of the American Federation of Musicians (A.F. of M.). Many organizations have negotiated their own agreements in recent years, so it is important that the producer consult with the orchestra management concerning the specific rules. In Europe the rules are different again, but in general there will be set record times and break times, as well as a limit on how long the orchestra can be asked to play in one sitting.

Additionally, there are limits on how much recorded material can be used from each session. This should be carefully assessed, since it will determine the minimum number of sessions that need to be booked for each project, thereby affecting the overall budget. Tuning is normally performed on record time rather than break time, except for the first tuning of each session, which is done just before the session begins.

> The clock is always running, whether it be tracking record time or break time—the dual clock used to track time during competitive chess matches comes to mind. A certain amount of skill is required of the producer to manage session times, as the precious record time should not be wasted on playbacks or lengthy discussions.

Running playback sessions during the breaks is an efficient use of the time, since the conductor will need to hear at least some of what is being captured. This will require the producer to choreograph the timing of the breaks so that playbacks happen at the right times. For instance, after a series of complete takes and small inserts have been recorded to finish a piece or movement, it makes sense to record a complete take of the next movement, rather than calling a break. This way, the conductor will have new material to listen to during the break, and a few minutes for any discussion with the producer.

13.2 Producer/Engineer Relationship

The simple explanation of the division of roles is that the producer is in charge of the music, and the engineer is in charge of the sound, but in reality there is a great deal of overlap. In most cases, the "overlap" will be biased toward the producer being actively involved in the sound, more than the engineer offering their opinion on musical issues. Engineers need to accept this inequity as part of the job description and be ready to collaborate with the producer (and the artist, for that matter) in creating the ideal sound for each project.

> A team effort usually yields the best results according to the old adage "two heads are better than one", or in this case "four ears are better than two". A recording team that is focused on working together should be capable of realizing superior results on a consistent basis than any individual endeavor.

The producer and engineer should maintain a "clientele" relationship. It is important that the engineer address all the producer's concerns, since the producer is normally responsible for choosing the engineer in the first place. The engineer will need to be ready to allow the producer to get involved in microphone choices and placement, as well as the usual balance requests. Some producers will want to be involved more than others when it comes to decisions that affect the sound. By contrast, there are those that will leave technical issues entirely up to the engineer such as overall timbre, perspective, and balance. This might seem very liberating except that in the case of an unhappy artist, that same producer might not be any help in buffering the artist's frustrations or offering any suggestions on improving the sound, so that the engineer may be left to fend for themselves.

A more even division of roles will develop over the course of several projects, and as the engineer earns the trust of the producer, musical issues may begin to be discussed more openly. For example the producer might ask the engineer which performance of a particular passage they prefer. It is good practice for the engineer to wait until they are asked for their opinion when it comes to musical considerations.

There is a famous joke concerning music production, although certain members of the recording team may not appreciate the humor: "*Every recording is either well produced or poorly engineered"*. In other words, the producer will take all the credit while the engineer is obliged to accept all the blame, based on the traditional hierarchy of the two roles.

13.3 Artist/Producer Relationship

While production styles may vary greatly between individuals, the most rudimentary function of the recording producer is to recognize and support the artist's vision for a particular performance, subsequently guiding them through the recording process. The description of "artist" extends to the conductor of an orchestra, a soloist, or a group of musicians such as a string quartet. This can be a very delicate process, and the producer needs to be well aware of the artist's personality and emotional state. It can be difficult for anyone with a strong musical background to "suppress" their own musical taste in this way, but may become necessary during the course of a recording session.

The producer should be very careful not to transform an artist's vision. There may be times when there is disagreement over performance practices, and the producer may point this out, but any proposed changes affecting musical style should be presented to the artist as suggestions or questions rather than explicit directions. It is important that the final interpretation of the recorded work is that of the artist and not the producer. One simply has to get "on board" with the artist, and help them to best present their version of the piece. After all, an artist that feels "bullied" rather than "guided" through the recording process might decide to look for a different producer for their next project.

The artist's state of mind needs to be constantly monitored, and diplomacy during all communications will be the key to a successful recording. A great deal of soul-searching and self-evaluation may be happening during the recording session, and the producer needs to help keep the artist focused on the task at hand, rather than dwelling on any negative aspects of the process.

One underrated skill all producers need to master is "talkback etiquette". This goes beyond the task of simply being sure the talkback microphone switch is depressed before speaking to the musicians on stage, and that the switch is released when the control room discussion is not meant to be "broadcast" into the hall. The all-important function of maintaining a constant dialog with the artist while they are out on stage or in the studio will help to ensure that the artist stays focused and feels supported by those in the control room. Certainly the communications will need to be concise during the time constraints of a union orchestral session, but the producer should at least say something positive after each take. This is even more important when recording smaller groups or solo artists. A musician who is sitting in an empty hall after performing a complete take of a piece will risk feeling completely unsure of themselves, unless a producer's encouraging words are heard immediately after the final chords have finished their decay.

13.4 Pre-production and the Role of the Producer

In order to be properly prepared for a recording project, the producer will need to be as familiar with the music as possible. Studying several existing recordings of the same piece is vital for the producer to have a sense of the range of available interpretations. This involves tempos, dynamics, and balances between instruments or sections of the orchestra. In the case of a previously unreleased work or a new composition, the producer should ask for a rehearsal recording or recent broadcast of a live performance if one exists. At the very least, a computer generated demo of the work might be used to help with learning the music. Knowing the rough timings of each movement or each piece can help in planning the recording session times.

Studying a piece ahead of time can also help the producer (and engineer) plan how they might want to position the ensemble for the best balance and overall presentation. This may include where certain instruments are placed on stage, so that they appear appropriately across the recorded soundstage.

Obtaining a copy of the score early on will ensure there is enough time to add markings and to follow the score while listening to other recordings of the work. The more familiar the producer is with the written page, the more attention can be paid to the actual music. While wrong notes and misplaced phrases are usually obvious artifacts, missing notes or parts that are not played are less apparent in the context of a recording session. The producer cannot always rely on the conductor or the individual musicians of a group to confess to those times when they have left out a few notes. These small details require great concentration on the part of the producer, and a substantial amount of preparation is necessary to be ready for the recording session.

Hearing the artist play the pieces before the recording can be very informative. If possible, attending a concert or a rehearsal can help greatly in preparing for the challenges of the recording session.

13.5 Running a Recording Session

In orchestral recording, the producer is normally tasked with running the recording session. A basic plan of approach is discussed with the conductor ahead of time, and usually each piece or movement of a symphony would be played several times, after which some smaller passages might be repeated as needed, to cover any places that were not performed well in any of the complete takes. Some conductors like to record by stopping whenever they hear a problem, and restarting with some overlapping material to fix the measures in question. In this case, one complete take would normally be recorded beforehand, in order for everyone involved to get an overview of the work, and for the producer and conductor to evaluate a complete performance while the orchestra is on a break.

The producer will be juggling many tasks during the course of the session, but the one underlying concern will be finishing the required amount of material before the end of the session. Constantly checking the clock, the producer has to prioritize their list of last-minute fixes to be recorded in the remaining minutes, saving the least important fixes for the very end of the session. Any technical details in the performance will need to be documented and reported to the musicians, with an appropriate amount of diplomacy. Through copious note-taking during the session, the producer will be able to tell the conductor and musicians with confidence that by combining the best segments of the recorded "takes", a satisfactory final version can be created in the editing process.

For chamber music or soloist projects that tend to run "off the clock", the producer may defer to the artist(s) in terms of directing the flow of events, while suggesting a playback or a break from time to time. In this case their role is more focused on making sure all of the material is properly covered across the takes, and that a perfect performance can be edited together based on the series of takes performed. There may be more input expected from the producer between takes, when time is less of a factor.

13.6 Post Production

After the recording sessions are complete, the producer is tasked with creating an edit plan based on notes from the session and comments from the artist. This is done in several ways, but most commonly markings are placed in the score showing the editor which performance or "take" to use at which time. Some artists like to "map out" the edits themselves, but normally the producer is expected to decide the edit plan with the artist's best interests in mind. In most cases, the marked score will be given to an engineer who will realize the edit by following the producer's map in the marked score.

In the case of a multi-microphone recording that needs refined balancing, a mix session will follow the editing phase. The producer and engineer will normally meet and mix together as a team, although it is assumed that the producer will direct most of the balance decisions based on musical reasons. At this point the producer is responsible for obtaining the artist's approval by incorporating edit and mix revisions requested by the artists, based on the various discussions with the conductor, soloist, and so forth.

13.7 Chapter Summary

- The producer is normally responsible for overseeing all stages of the recording process—including the performance and the overall sound.
- A producer needs to have a comprehensive understanding of both music and recorded sound.
- Producers normally run the recording session, and keep track of all the recorded takes of each piece, plan the editing map, and direct the mixing session.