3
Ideal and Technical Filters

Whether implicitly or explicitly, each of the musical examples I have described in the previous chapters addresses the issue of time. As the aesthetic organization of sound waves that take place *over time*, music is by definition a temporal art form. Indeed, the temporality of sound has always been one of the pillars of musical creativity. As this book is centrally concerned to point out, though, technological sound reproduction introduced an inherent paradox: while it allows sound flows to be replicated almost identically, again and again, the procedures of capturing and replaying themselves also shape the sounds in the process. As both digitally captured and infinitely repeatable sonic documents *and* records of irreversible material decay, Basinski's *The Disintegration Loops* encapsulates this contradiction. The musical substance of these pieces lies not in the initial loops, but rather in how the disintegrating magnetic coating affected, changed, and shaped their sonic contours in random, unexpected ways. Each time *The Disintegration Loops* are replayed—that is, each time the digital data are transduced back into electrical currents and then sound waves—this singular process of decay is heard again and again and again.

Similarly, The Caretaker's *Patience (After Sebald)* turns the surface noise, needle scratch, crackles, hisses, and hums of an old Schubert recording into the main musical event. In a sonic accompaniment to W.G. Sebald's literary meditations on time, memory, and melancholia, the piece emphasizes the unbridgeable distance between us as listeners and the faint, muffled, and distant voice and piano on the old recording. These come to us, at the thresholds of recognition, in layers of crackle and noise. Even the twenty seconds of pure tape noise that kick off Mark Hollis's solo album makes the listener acutely aware that they are listening to a recording—to music that was committed to magnetic tape on a specific day in a specific room. Simultaneously with this, the noise creates a sense of intimacy that is not unlike "listening in" to a private phone call. In this way, "The Colour of Spring" seems to transport listeners back to the moment, sometime in the past, at which the music was recorded. We do not just hear Hollis sing, we hear him sing *in a microphone*, or rather, given that we are on the receiving end of the line, we hear him sing *through* a microphone and the long chain of further machines connecting us

The Logic of Filtering. Melle Jan Kromhout, Oxford University Press (2021). © Oxford University Press.
DOI: 10.1093/oso/9780190070137.003.0004

with him. We pick up on the sonic markers indicating that technology has shaped these sounds, confirming that we listen not only *through* noise as it mixes with the signal and shapes it, but also *to* it.

The fact that noise and distortion are important in how we make sense of recorded sound is a result of what I call the noise resonance of sound media, which serves as a counterpart or conceptual foil to the myth of perfect fidelity. Based on a conceptual logic of noise reduction, this myth originated in the early days of recording technology before being consolidated by communication engineers in the 1920s and '30s and information theory in the 1940s and '50s. Although the conceptual logic of noise reduction assumes that it is possible to cleanly remove, suppress, or reduce noise, the previous chapter showed how dual-ended noise reduction systems only reduce what has already been framed as "noise," and always affect the signal in the process. What is more, when it comes to digital systems, the reintroduction of noise in the form of dither to alleviate digitization's structural limitations actually turns noise into a positive, maybe even fundamental element of the recording process. This recognition and reemergence of noise as a structural necessity problematizes the conceptual logic of noise reduction. In response to this, the idea of a noise resonance of sound reproduction disregards the myth of perfect fidelity. Instead of denying and repressing noise, it acknowledges the crucial roles played by noise, distortion, and randomness for how listeners make sense of technologically reproduced sound and music.

The practice of dithering in particular accentuates the gap between conceptual ideals of perfect reproduction and their physical implementation in technical media. Like the random noise produced by analog recording technologies, the quantization errors and aliasing effects of digitization highlight sound reproduction technologies' fundamental limitations. They conform that, despite the many differences between them, analog and digital media ultimately adhere to the same general logic of signal transmission. Fleshing out the conceptual implications of a noise resonance of sound media in more detail, then, requires looking beyond the analog/digital divide. In this chapter, I therefore trace some of the historical and theoretical (mathematical, physical, and discursive) foundations of technological sound reproduction. And as we will see, the theoretical principles of sound reproduction involve questions of temporality with which musical practice has also long been concerned.

The first half of this chapter broaches the significance of time in relation to two physical uncertainty principles fundamental to communication engineering and information theory. Following Abraham Moles's classic book *Information Theory and Esthetic Perception*, I show how these principles entail compromises that limit the accuracy of any reproduction or representation, because they require a tradeoff, first, between maximum dynamic range and maximum frequency range, and, second, between dynamic range and temporal accuracy. In this way, they provide a physical foundation for Shannon's conceptualization of noise as both a prerequisite for, and disturbance of, successful signal transmission.[1]

Then, moving into the second half of the chapter, I trace the limitations imposed by these uncertainty principles back further still, to mathematical Fourier analysis in the early nineteenth century. As I described in the previous chapter, Fourier analysis is a powerful analytical tool that can be used to represent any sound spectrum as a series of sine waves—elemental sounds with a single frequency. Crucially, in idealizing physical sound, Fourier analysis symbolically does away with noise *and* time altogether. The inevitable introduction of noise in physical sound reproduction, then, marks a fundamental gap between the idealized, mathematical representations that underpin modern models of physical sound and sound media, and physical processes of technological reproduction that unfold in space *and* over time.

The Limits of Representation

According to the myth of perfect fidelity, the ultimate goal of technological sound reproduction is to produce perfect copies. Any distorting or disrupting effect should be prevented, reduced, or eliminated, for only when all random, transient, and unexpected events are expelled will we have complete control over the process of reproduction. Our capacities for attaining such absolute control and precision are fundamentally restricted, however, because the reproduction and representation of signals require compromises between incompatible extremes at the most elementary physical level. Like Werner Heisenberg's well-known uncertainty principle in quantum mechanics, formulated in 1927, the compromises or trade-offs entailed by two closely related uncertainty principles in signal processing limit the possible accuracy of any system.[2] The first principle describes a relation of uncertainty between, on the

[1] Moles, *Information Theory*, 5, 83–87.

[2] Heisenberg's uncertainty principle in quantum mechanics states that a subatomic particle cannot simultaneously have an exact location and exact momentum, which means that if we observe *where* the particle is, we do not exactly know *when* it is at that position. Conversely, if we know *when* the particle is observed, we cannot tell exactly *where* it is located. Regarding the relation between Heisenberg's uncertainty principle and the uncertainty principles in communication engineering and information theory discussed here, Kosko points out that even before the publication of Heisenberg's uncertainty principle in 1927, "versions of this uncertainty trade-off appeared at Bell Laboratories and elsewhere" in the context of signal processing. Kosko, *Noise*, 113. Indeed, Norbert Wiener shows that both Heisenberg's uncertainty principle and the uncertainty principles in communication engineering and information theory can be explained "through the same harmonic analysis." Norbert Wiener, "Spatio-Temporal Continuity,

one hand, the system's sensitivity to very faint (low-amplitude) signals and, on the other, the width of its frequency spectrum (its sensitivity to very high and very low sounds). The second indicates the relation between the system's sensitivity to low-amplitude signals and its sensitivity to a signal's duration.[3] In each case, as soon as the accuracy in one domain increases, it decreases in another; and vice versa. In this sense, both uncertainty principles require a trade-off between irreconcilable poles.

In the first trade-off (between minimum signal amplitude and maximum frequency range), the narrower the frequency bandwidth of a transmission channel, the more sensitive it will be to low-amplitude signals. Conversely, the broader a channel's bandwidth, the less sensitive it will be to low-amplitude signals. Thus, a signal with a broad frequency range will have a smaller dynamic range, whereas a system with a large dynamic range will have a narrower frequency range. The cause of this conundrum is the presence of noise. Despite information theory's premise that noise and signals can be separated according to randomness (noise) and periodicity (signal), very faint signals are sometimes drowned out by background noise (the noise floor). Paraphrasing Einstein, Moles writes that this "background noise is due to the agitation of electrons in conductors," meaning that it is present down to the level of elementary particles.[4]

To prevent signals from being drowned out by noise, their energy level can be raised—by amplification, for instance. However, given that this would amplify the noise floor too, all amplification eventually reaches a point at which noise will overtake signal entirely, rendering further amplification useless.[5] Alternatively, one can minimize background noise by cooling the equipment. Because the agitation of electrons is proportionate to temperature, there will be no movement of electrons and no noise at absolute thermodynamic zero. In practice, however, it is only possible to lower the temperature of audio equipment within reasonable, fairly restricted limits.

Narrowing the bandwidth of a channel presents a more practical way of increasing a system's sensitivity to low amplitudes. It focuses its capacity on an increasingly small band of frequencies and filters out more and more

Quantum Theory and Music," in *The Concepts of Space and Time: Their Structure and Their Development*, ed. Milič Čapek (Dordrecht: D. Reidel Publishing Company, 1976), 545–546. Even more so, mathematician Gerald Kaiser argues that, "contrary to some popular opinion," the uncertainty principle "is a general property of functions, not at all restricted to quantum mechanics." Gerald Kaiser, *A Friendly Guide to Wavelets* (Boston: Birkhäuser, 1994), 52.

[3] Moles, *Information Theory*, 83–87.
[4] Moles, *Information Theory*, 84.
[5] Moles, *Information Theory,* 85.

noise. Still, gradually narrowing a filter's focus increases the risk that it will also filter out frequencies belonging to the signal itself. For example, imagine removing the noise from a 78-rpm recording of an orchestral symphony. The frequency range of a symphony orchestra (like that of most music) is generally quite broad. Accordingly, filtering out parts of the spectrum that are most heavily affected by "unwanted" noise will almost certainly also affect some higher and lower frequencies belonging to the "wanted" signal. As the filter narrows, increasingly large portions of music are lost, up to the point where essential musical information—the low rumble of timpani, for instance, or the high-pitched string section—disappears. Hence, a smaller frequency range increases sensitivity toward low-amplitude signals, but decreases the frequency range. The risk here, then, is that of throwing out the baby (signal) out with the bathwater (noise).

The symbolic limit case—the most extreme, idealized manifestation—of this conundrum would be an ideal noise filter that narrows bandwidth to just a single frequency, blocking or filtering out all frequencies but one, irrespective of whether they are deemed noise or signal.[6] By that point, all characteristics of the signal, in terms of its unique frequency spectrum and development over time, would have disappeared. The music's spectral richness would be reduced to a single frequency, which would convey just as little information as the noise in the amplification scenario. Indeed, although such a filter would confirm that a signal is there, all of its spectral and temporal specificities would have been stripped away, leaving no way of knowing what the signal was. This single frequency would be entirely noiseless, but all information about what kind of message was transmitted (is it symphonic music, a speech, rock performance, or field recording?) would be lost. This signal would only provide a single bit of information: that it is either present or not.[7]

Hence, neither signal amplification nor noise reduction gets around having to find a tradeoff between sensitivity to low-amplitude signals and reproducing a broader frequency range. Where no filter is applied, the frequency band is limited by physical background noise, which drowns out all signals below a minimum amplitude threshold. Although the energy level of low amplitude signals can be amplified, too much amplification will raise the noise floor to a level at which it threatens to overtake the signal entirely. Conversely, installing a noise filter to narrow the frequency spectrum and reduce the noise floor introduces the risk of losing part of the signal itself. Taken to an extreme, only a single frequency would remain. The first uncertainty principle

[6] Moles, *Information Theory*, 87.
[7] Moles, *Information Theory*, 85.

Box 3.1 The First Uncertainty Principle

- Abraham Moles: "Error in amplitude × error in frequencies = constant"[a]
- System is more sensitive to lower amplitudes = narrower bandwidth = less noise = narrower frequency spectrum
- Broader bandwidth = more frequencies transmitted = more noise = system less sensitive to lower amplitudes

[a] Moles, *Information Theory*, 85.

(as summarized in Text Box 3.1) thereby shows that the physical presence of noise down to the most elementary level fundamentally limits the maximum capacity of any transmission channel. The wider a channel's bandwidth, the more noise it admits and the less sensitive it is to low-amplitude signals; the greater its sensitivity to low-amplitude signals, the more noise (and eventually signal too) will be filtered out and the more the frequency spectrum will narrow.

The second uncertainty principle follows directly from the first. In addition to frequency spectrum and dynamic range, it also involves signal duration. As such, it further accentuates the physical limitations described previously. Moles introduces the relation between the two principles with a thought experiment, describing a device that seemingly solves the first trade-off between low-amplitude signals and frequency range.[8] Imagine a machine consisting of a great many ideal filters of the type described earlier, each solely attuned to a different, single frequency. Hypothetically, this machine would enable the transmission of all these frequencies without any interfering background noise. It is, however, both logically inconsistent and physically impossible.

First, determining which frequencies each of the filters should process requires unambiguous information about the signal's frequency spectrum and the noise that should be removed. And yet, this information is not available prior to the filtering operation itself: indeed, if it were, one would not need Moles's hypothetical machine to separate signal from noise. This means, Moles argues, that the problem of determining which frequencies belong to the signal and which to the noise remains, because the transmission of a complete signal would require infinitely many filters.[9] Second, and even more fundamentally, the very concept of a filter attuned to a single frequency is physically impossible. Because nothing in the world happens instantaneously,

[8] Moles, *Information Theory*, 87.
[9] Moles, *Information Theory*, 87.

Box 3.2 The Second Uncertainty Principle

- Abraham Moles: "Error in amplitude × error in duration = constant"[a]
- System is more sensitive to lower amplitudes = narrower bandwidth (= less noise) = longer delay = more uncertainty about duration
- Precise duration = shorter delay = broader bandwidth (= more noise) = system less sensitive to low amplitudes

[a] Moles, *Information Theory*, 87.

any real sound reproduction system will require a minimum amount of time to process a signal. This means that a filter requires a minimum response time to fulfill its task, slightly delaying the production of an output. As with the uncertainty relation between frequency response and sensitivity to low-amplitude signals, this delay is "proportional to the narrowness" of the filter.[10] A system with a narrower filter will exclude more frequencies, create longer delays, and take more time to produce an output. These negative effects will increase or decrease in proportion to the signal's bandwidth.

The frequency spectra of most signals are not stable and periodic, but rapidly and continuously changing. Indeed, they sometimes change faster than filters' minimum response time, causing inaccuracies in the output. If a frequency spectrum changes before its filter has completed its analysis, then this change will not be processed. As a result, the duration of the filtered frequencies is registered incorrectly. Hence, although a narrower channel will reduce more noise and transmit lower-amplitude signals, filtering out more frequencies requires more time, which causes a longer response time and longer delay. Following the second uncertainty principle (as summarized in Text Box 3.2), any gain in sensitivity to low-amplitude signals comes at the cost of sensitivity to their duration. In an extreme limit case of this trade-off, the aforementioned infinitely accurate filter would ideally filter out all noise so as to (re)produce just one, absolutely noise-free frequency. Significantly, in this instance, the filter's response time would mathematically tend toward infinity. Indeed, given that such a filter would never stop filtering, it would never produce its perfectly unambiguous output. Though mathematically perfect, the ideal filter is physically impossible.

Despite their physical impossibility, conceptualizing such ideal filters is an essential practice in theoretical physics and technical engineering, for doing so helps us understand the physical filtering operations of technical media. As

[10] Moles, *Information Theory*, 86.

Vilém Flusser argues, practices of modeling do not objectively represent the physical world, but order and structure it.[11] In a manner akin to Heidegger's account of "enframing" (explained in chapter 2), Flusser argues that the "so-called natural laws" of physics are not objective descriptions of physical processes but ways of decoding the "gigantic quantity of indications, signs, clues" with which we are confronted on a daily basis.[12] Hence, mathematical and physical models are not neutral representations, but idealized versions of complex processes. In symbolically establishing order and reducing complexity, they also shape our perspective on, and approach to, the phenomena they represent.[13] When our attempts to represent and reproduce physical processes run up against their physical complexity, idealized models serve to break up this complexity to impose order, regularity, and linearity. Without mathematical conceptualizations, and the symbolic understanding of otherwise ungraspable processes that they provide, technological development would be close to impossible.

Still, in physical reality, the seamless operations of an ideal filter are faced with the physical limitations imposed by the uncertainty principles described earlier. This applies to both digital and analog media, for even machines that can process signals as fast and accurately as modern digital media eventually run into the intractable physical constraints posed by the trade-offs I have outlined. In the case of digital recording, higher precision (more bits) decreases the number of quantization errors and allows a more precise representation of each sample's amplitude value. By reproducing low-amplitude signals more accurately, it enlarges the dynamic range. On the other side of the ledger, however, the uncertainty principle means that measuring and processing longer word lengths also requires longer response times and corresponding delays. This, in turn, introduces errors with respect to signal duration. Conversely, a higher sample rate increases frequency response and allows for a broader frequency range. Higher sample rates multiply the number of samples per second, however, meaning that they require shorter

[11] Vilém Flusser, *Into the Universe of Technical Images*, trans. Nancy Ann Roth (Minneapolis: University of Minnesota Press, 2011), 170.

[12] Flusser, *Universe*, 46.

[13] Flusser, *Universe*, 170. "Models," Flusser writes, "give form to a world and a consciousness that has disintegrated; they are meant to 'inform' that world. Their vector of signification is therefore the reverse of that of earlier images: they don't receive their meaning from outside but rather project meaning outward. They lend meaning to the absurd." Similarly, John Monk writes, "it is tempting to imagine that a model or theory is an accurate reflection of what takes place in reality; however, prominent nineteenth century physicists and latterly pragmatist philosophers have insisted that our descriptions of reality are of our own making and are a product of our institutions and customs. Models as part of our descriptive practices, therefore, make a contribution to the construction of reality." John Monk, "Creating Reality," in *Ways of Thinking, Ways of Seeing: Mathematical and Other Modelling in Engineering and Technology*, eds. Chris Bissell and Chris Dillon (Berlin: Springer, 2012), 2.

samples with shorter response times. Ultimately, this decreases the system's sensitivity to low-amplitude signals.

The fact that both analog background noise and digital quantization errors limit a system's sensitivity to low-amplitude signals can thus be explained in terms of these uncertainty principles. This means that the most fundamental difference between ideal models and the physical systems they represent is the presence of noise: random physical noise in the case of analog media, and communicational noise (error and distortion) in the case of digital media. The uncertainty principles mark the gap between the idealized domain of mathematical models, which I will call the plane of the ideal filter, and the operations of filters in physical reality, which I will call the domain of technical filters.

As the previous chapter explained, this gap results from what Siegert calls the rift or rupture in the classical order of representation, which appeared with the development of modern mathematical analysis between Leibniz's invention of the infinitesimal calculus in the late sixteenth century and the emergence of Fourier analysis in the early nineteenth century. The introduction of Fourier analysis in particular allowed for the symbolic representation of complex physical processes and eventually their autonomous reproduction by technical media. A crucial moment in the growing divide between representation and represented, Fourier analysis marks the beginning of a new order, built on the twin pillars of analytical idealization and physical reproduction. Conceptually, the domain of the ideal filter is grounded in Fourier's analytical theorem, which also produced the idea of the sine wave as one pure frequency. The application of Fourier's theorem to the analysis of sound, in short, inspired the conceptual logic of noise reduction's ideals of infinite precision and maximal purity (although it should be said that the mathematics underpinning the uncertainty principles described earlier can be traced back to Fourier's work as well). A closer look at the history and basic principles of Fourier analysis and the figure of the sine wave, therefore, can help explain the relation between the noiseless plane of the ideal filter and the importance of noise in the domain of technical filters.

Fourier Analysis: A ShortIntroduction

First published in 1822, Fourier analysis became a crucially important analytical tool and cornerstone of our contemporary information society.[14] At

[14] As mathematician T. W. Körner puts it, Fourier analysis is "built into the commonsense of our society." Körner in Barbara Burke Hubbard, *The World According to Wavelets: The Story of a Mathematical Technique in the Making* (Wellesley, MA: A K Peters, 1996), 8. Also see Donner, "Fourier's Beitrag."

the time of its emergence, it marked a decisive step in the transition from a received type of physical research, for which mathematical analysis was of secondary import, toward modern theoretical physics, for which mathematical modeling is an absolutely central, indeed constitutive practice.[15] Jean-Baptiste Joseph Fourier had been a prodigious mathematician from early childhood. Later, his experiences under the hot desert sun of Egypt, where he served in Napoleon's armies during the final years of the eighteenth century, allegedly provided him with a lifelong obsession with heat. As his biographer John Herivel writes, Fourier was unable "to acclimatize himself to the change from Egypt."[16] Upon his return to France in 1801 or 1802, "the question of heat, its loss by propagation in solids and radiation in space, the problem of conserving it [. . .], can never have been out of his mind for long."[17] From 1804 at the latest, Fourier—whom Napoleon had by now appointed prefect of the newly created Département Isère in Grenoble—was spending most of his free time developing a new theory of heat propagation.

When the first version of his treatise "On the Propagation of Heat in Solid Bodies" was completed in 1807, it was met with considerable resistance from leading physicists of the time, most notably Siméon Poisson, Jean-Baptiste Biot, and Pierre-Simon Laplace. In 1811, Fourier entered a revised version of the treatise into the contest for that year's grand price in mathematics at the Institut de France. Not entirely by coincidence, the topic of the competition—"the propagation of heat in solid bodies"—matched his interests exactly. After amending, correcting, and expanding on the first draft, he won the contest. Still, the committee remained convinced that his solutions were "not exempt of difficulties" and left "something to be desired."[18] Due to this persistent professional opposition, and his ongoing, turbulent political career, the definitive version of the *Analytical Theory of Heat* was only published in 1822. This was after Fourier had been appointed permanent secretary for the mathematical sciences at the Académie des Sciences in Paris and eight years before his death in 1830.

Much of the professional resistance that Fourier had faced in the fifteen years between his draft of 1807 and the final treatise in 1822 was due to his scientific approach. Rather than relying on qualitative results based on empirical research, Fourier extrapolated from empirical findings to develop new hypotheses based on advanced mathematics. Steeped in the work of late eighteenth-century mathematicians and physicists, Fourier was especially

[15] Olivier Darrigol, "The Acoustic Origins of Harmonic Analysis," in *Archive for History of Exact Sciences* 61, no. 4 (2007): 397.

[16] John Herivel, *Joseph Fourier. The Man & the Physicist* (Oxford: Clarendon Press, 1975), 99.

[17] Herivel, *Fourier*, 99.

[18] Committee report cited in Herivel, *Fourier*, 103.

good, writes Herivel, at "taking an essentially complex problem and make it amenable to mathematical treatment while simultaneously providing a solution yielding a good approximation to the actual physical situation in a wide range of cases."[19] Indeed, his work introduced a level of mathematical abstraction that, although still highly contested during the late eighteenth and early nineteenth centuries, was to become standard practice over the course of the nineteenth century.

Mathematically, Fourier analysis transforms a function (*f*), representing development over time, into a series of sine and cosine values corresponding to partial states. In the case of sound signals, these values correspond to the amplitude, phase, and frequency of every individual wave in its sound spectrum. The outcome of this transformation is a mathematical representation of the waveform, which is given in terms of a frequency spectrum consisting of many sinusoidal components called "sine waves." The sum of all these frequency components expresses the original waveform. This is a Fourier series, the simplest rendition of the Fourier transform, which applies solely to periodic signals that repeat identically over and over again.[20] The frequency composition of such signals can be complex (in that they consist of many individual sine waves oscillating at different frequencies). In terms of its temporal development, however, every cycle of a periodic signal is exactly the same. Given this periodicity, one cycle contains all available spectral and temporal information about the signal. This means that only one cycle is needed to analyze its frequency spectrum. The time required to analyze a strictly periodic signal is therefore identical with one cycle's duration.[21]

The requirement that analysis focus solely on endlessly repeating periodic signals follows from Fourier's mathematical use of trigonometric or circular sine and cosine functions.[22] These functions derive from the representation of the ratios between the sides of a triangle (hence the "trigonometric" in their name), which stems, in turn, from the geometrical relations between triangles and circles (hence the name "circular"). The origins of Fourier analysis in the geometry of circles is significant on account of the fact that a circle's circumference is mathematically infinite. In accordance with this, the infinity of sinusoidal motion is an analytical given in Fourier analysis.[23] To summarize: in

[19] Herivel, *Fourier*, 213.
[20] Most textbooks write that the Fourier *series* applies to periodic functions and the Fourier *transform* to nonperiodic or quasi-periodic functions. Engineer Stan Tempelaars, however, argues that "the Fourier series [. . .] is the Fourier transform of a periodic function," which means that a Fourier series is a particular form of the more general Fourier transform. Stan Tempelaars. *Signal Processing, Speech and Music* (Lisse: Swets & Zeitlinger Publishers, 1996), 142.
[21] Tempelaars, *Signal Processing*, 129.
[22] Gareth Loy, *Musimathics: The Mathematical Foundations of Music*, Volume 1 (Cambridge, MA: MIT Press, 2006), 140.
[23] Loy, *Musimathics*, 140.

the symbolic domain of Fourier analysis, sine waves are infinite by definition. A Fourier series represents a periodic signal, which consists of series of sine waves, as if it has oscillated and will oscillate, unchanged, for all eternity.

The infinite repetition of sine waves on the mathematical plane poses few problems for the analysis of periodic signals. Their periodicity already entails their endless repetition (at least in theory). Nonperiodic signals, by contrast, change over time, often rapidly. To deal with this changeability, the analysis of nonperiodic signals employs a trick. Instead of simply representing the original waveform as the sum of all individual sine and cosine values, the Fourier transform of nonperiodic signals applies a second idealization. It treats a nonperiodic signal as if it were periodic, approaching the entire signal (or some part of it) as one complete cycle of an imaginary periodic signal. To achieve this, the analysis assumes that the temporal factor (t), which represents one full cycle, is infinitely long. Effectively, this means that the Fourier transform of nonperiodic signals renders irrelevant time conceived as "duration" (the time of things with a beginning and end).[24]

Then, to derive the frequency spectrum of this (symbolically) infinite cycle, the Fourier transform replaces the sum of all sine and cosine values with an integral. This means it adds up or "integrates" a great number of little (ideally infinitesimal) slices of the "cycle" to represent all of its frequency components. By essentially pretending that the nonperiodic signal is periodic, the Fourier transform thereby represents all the sine and cosine values as one artificial, infinite "cycle."[25] In short, as the pioneering engineer Ralf Heartley puts it, the Fourier integral "may be thought of as a mathematical fiction for expressing a transient phenomenon in terms of steady state phenomena."[26] In this context, this means the representation of unpredictable, nonperiodic wave phenomena that constantly change over time as periodic, regular signals that repeat infinitely and unchanged.

At the expense of the Leibnizian ideal of complete representability and absolute correspondence between mathematical representation and physical reality, the analytical idealizations introduced by the Fourier transform create a perspective from which all ambiguity is expelled, and perfect clarity seems to appear. Although these idealizations closely approximate the properties of the physical phenomenon they represent, the correspondence is never exact.

[24] Tempelaars, *Signal Processing*, 129.
[25] Tempelaars, *Signal Processing*, 129.
[26] Heartley in Wolfgang Ernst, *Chronopoetik: Zeitwesen und Zeitgaben Technischer Medien* (Berlin: Kadmos, 2012), 40. William Sethares defines "steady state" as "the part of a sound that can be closely approximated by a periodic waveform" and a "transient" as "that portion of a sound that cannot be closely approximated by a periodic signal." William A. Sethares, *Tuning, Timbre, Spectrum, Scale*, Second edition (London: Springer, 2005), xviii.

According to Siegert, the graphs depicting Fourier's analysis of heat propagation represent "the seemingly sharp contours of surfaces, which are actually just the infinitely fine heat-shimmering of these surfaces themselves."[27] What appears in all sharpness and clarity in the Fourier domain is actually an idealized representation of constantly changing transient phenomena in an artificial "steady state."

Fourier himself did not apply his analytical method to acoustical problems. Although his work drew heavily on an eighteenth-century controversy regarding the mathematical representation of vibrating strings, it was only in the 1840s and '50s that Georg Simon Ohm and Hermann von Helmholtz applied Fourier analysis to the study of musical sounds.[28] Ohm turned to Fourier's treatise in 1843 to prove the hypothesis that complex sound waves can be represented by series of these "simple" waves. Unlike acoustician August Seebeck, who heavily objected to this abstract mathematical approach, Ohm did not rely on empirical observations using his own faculty of hearing. Instead, as Julia Kursell argues, "mathematics [. . .] replaced the ear for Ohm," who "sought to illuminate physical phenomena with the help of their mathematical formalization."[29] Notwithstanding Seebeck's strong objections, which caused Ohm to retreat from the field of acoustics altogether, Helmholtz later corroborated, corrected, and expanded Ohm's analysis. *On the Sensations of Tone as a Physiological Basis for the Theory of Music*, Helmholtz's highly influential work on the nature of sound and the physiology of hearing, was first published in 1863. By combining Ohm's mathematical analysis with extensive empirical experiments of his own, the book established Fourier analysis as the quintessential theory of the composition of sound waves. Building on what came to be known as "Ohm's Acoustic Law," Helmholtz even argued that the ear itself performs some kind of Fourier analysis.[30]

Helmholtz set out to verify the mathematical outcomes of the Fourier analysis of periodic tones (or "musical" tones, as he has it) through scientific experiment and prove empirically the physical difference between these and nonmusical sounds (in other words, noise) through empirical observation. In these efforts, the strict periodicity assumed by Fourier analysis encouraged Helmholtz to take, as Kursell puts it, "the steady, internal repetition of

[27] Siegert, *Passage*, 246.

[28] See Georg Simon Ohm, "On the Definition of a Tone with the Associated Theory of the Siren and Similar Sound Producing Devices," trans. R. Bruce Lindsay, in *Acoustics: Historical and Philosophical Development*, ed. R. Bruce Lindsay (Stroudsberg: Dowden, Hutchinson & Ross, 1972), 242–247; Helmholtz, *Sensations*, 84–100; R. Steven Turner, "The Ohm–Seebeck Dispute, Hermann Von Helmholtz, and the Origins of Physiological Acoustics," *British Journal for the History of Science* 10, no. 1 (1977), 1–24.

[29] Julia Kursell, "Experiments on Tone Color in Music and Acoustics: Helmholtz, Schoenberg, and Klangfarbenmelodie," in *Osiris* 28, no. 1 (2013): 196.

[30] Turner, "Ohm–Seebeck Dispute," 5.

periodic sound waves" as the starting point of his experiments, and approximate "the mathematical description of a periodic wave as closely as possible."[31] This means that he tried to come as close as possible to producing absolutely pure sine waves. To produce the most periodic sounds that he could physically achieve, Helmholtz built an experimental set-up consisting of tuning forks fitted with resonating cones to amplify their basic frequency. These artificial sine-like sounds effectively constituted a new type of sound, for which no physical referent had hitherto existed.[32]

In this way, Helmholtz forged a connection between the mathematical idealizations of Fourier analysis and the acoustic phenomena under investigation. The strictly periodic and ideally infinite sine wave, which Fourier himself never explicitly mentions, was not so much the object of Helmholtz's experimental analysis as its product. No longer a purely mathematical concept, the sine wave was now also an acoustic object produced to approximate that mathematical ideal. As such, it presented an empirical basis for age-old discursive connections between music, harmony, and regularity that go back as far as Pythagoras's theories of celestial harmony in the sixth century BC.[33] The mathematical-acoustic figure of the infinite sine wave also constitutes a foundational element in the ideal of a perfect, noiseless signal. Indeed, it would become the quintessential figure on the plane of the ideal filter that was to dominate the discourse on sound and media from then onward.

The Plane of the Ideal Filter

With Ohm's application of Fourier's theorem to the analysis of sound, and Helmholtz's expansion and experimental verification of its principles, the sine wave came to be defined as the elemental tone: a pure frequency with no overtones and no timbral characteristics of its own. As with the graphs showing the "seemingly sharp contours of surfaces" produced by Fourier's analysis of heat propagation, the purity and clarity of the sine wave are properties of a conceptual limit case—the most extreme limit of a phenomenon—drawing

[31] Kursell, "Experiments," 192, 205.
[32] Kursell, "Experiments," 192.
[33] As Douglas Kahn writes, "the figure of vibration was upheld by the Pythagoreans, refurbished by neo-Platonic and neo-Pythagorean thought centuries later, and invigorated by scientific, Eastern and spiritist thought in the West in the nineteenth century. The monochord—the technology that underscored the harmonic totality of Pythagorean thought, the vibrating string structuring the cosmos—was so overcoded by the late-nineteenth century locus of vibrations in the synesthetic arts that it was functionally nonexistent, although the connections between acoustics, music, and mathematics, not to mention certain ambitions toward the cosmos, remained strong." Douglas Kahn, *Noise, Water, Meat: A History of Sound in the Arts* (Cambridge, MA: MIT Press, 2002), 16.

seemingly sharp contours around infinitesimally fuzzy sound waves. A fixed and indivisible standard of sonic purity, the sine wave is an ideal form toward which all physical sounds seem to tend. The relation between this idealized mathematical object (a sine wave) and the physical phenomenon it represents (a simple sound wave) can therefore be described as the relation between symbol and signal.[34]

Produced by the strictly symbolic operations of mathematical analysis, a sine wave is not a physical signal, but an analytical symbol. Its symbolic clarity depends upon a prior, conceptual act of noise reduction that suppresses all reference to its material carriers (transmission channels). "The mathematician," Serres explains, "does not see any difficulty on this point," for the mathematical manipulation of written signs already serves "to isolate an ideal form [and] render it independent of the empirical domain and of noise."[35] The mathematical production of an ideal symbol, in other words, entails the removal of any trace of its material production as signal. Ultimately, this entails denying its physical production and transmission as signal, and thus the complete symbolic reduction of noise. To function mathematically, the sine wave requires a process of abstraction that separates its "pure" symbol from its physicality as a contingent signal. In subsequently being physically produced as an actual acoustic object, the sine wave becomes what we might call an "idealized signal," discursively positioned in between purely symbolic mathematical analysis (the plane of the ideal filter) and physical acoustics (the domain of technical filters).

So, the production of a perfect sine wave—a single frequency—would require complete noise reduction. This indicates that Fourier analysis and the concept of the sine wave are subject to an uncertainty principle. The ideal sine wave presupposes the analytical filtering out of all material channels; and because it represents a signal as a series of such sine waves, the operations of Fourier analysis can be interpreted as an ideal—that is, infinitely accurate—spectral filter.[36] Following the uncertainty principle described earlier, the more a physical filter comes to resemble this ideal filter, the narrower its frequency range becomes and the more time it will need to complete the operation. At the analytical limit of this process, the filter will be attuned to a

[34] Siegert, *Techniques*, 19–23.

[35] Michel Serres, *Hermes: Literature, Science, Philosophy*, trans. and eds. Josue V. Harari and David F. Bell (Baltimore: Johns Hopkins University Press, 1982), 68, 70.

[36] Referring this ideal filtering operation back to its origins in Helmholtz acoustical experiments, Tara Rodgers writes, "the technical process of regulating additional harmonic frequencies is now known as filtering—which retains Helmholtz's logic of separating out the pure form from the flux of variations." Tara S. Rodgers, "Synthesizing Sound: Metaphor in Audio-Technical Discourse and Synthesis History" (PhD diss., McGill University, Montreal, 2010), 120.

single frequency and its response time will tend mathematically to infinity. At that point, the physical filter would become an ideal filter and the physical signal an ideal signal: a pure sine wave. In this way, the infinity of the sine wave correlates directly with the uncertainty principle: an ideal filter produces a single, symbolic frequency only when the factor t (in mathematical terms) or the filter's response time (in engineering terms) is infinite.

Kittler explains this correlation between the mathematical idealizations of Fourier analysis and the abstraction from temporality through the metaphor of lightning and thunder. An (ideally infinitesimally) short event (lightning), he emphasizes, can be analyzed in terms of a series (thunder).[37] On account of its briefness, the only information that one can generally glean from a lightning bolt is its "thatness" (*dass es ist*)—the simple fact that it took place. Understanding its "whatness" (*was es ist*), however, requires that the event repeats itself, that one stretch it out in time so as to allow assessment and analysis.[38] In the case of lightning, such repetition comes in the form of the acoustic reverberations of thunder. Repetition (or rather "frequentia, the return") provides time in which to analyze the singular event and acquire more stable knowledge about what happened.[39] This, Kittler argues, is what Fourier analysis does: it transforms a brief, random, and constantly changing signal into a series of repetitions or frequencies. As thunder is to lightning, so the frequency domain created by Fourier analysis is to the original signal.

The Fourier transform imposes a temporal (one could even call it rhythmical) order in the form of an infinite, periodic repetition of simple elements. This repetition allows waves that unfold in time to be taken out of time. The Fourier domain—the plane of the ideal filter—conjures a fundamentally atemporal sphere in which everything always returns and nothing ever changes. As a perfectly noiseless signal, the sine wave exemplifies this abstract plane of existence. As I have set out earlier, symbolic representations of the sine wave imply the complete exclusion of material channels. Physically, totally transcending channels in this way would require the complete reduction of noise—and that, in turn, would symbolically require removing the factor of time. The actually existing domain of technical filters, by contrast, is defined by the impossibility of such analytical purity, by the occurrence of events that physically change over time and inevitably introduce a level of random noise. It is marked not by the frequentia of the series, but by the flash of the event.

[37] Friedrich Kittler, "Lightning and Series—Event and Thunder," Theory, Culture & Society 23, no. 7–8 (2006): 63–74.
[38] Kittler, "Lightning," 70.
[39] Kittler, "Lightning," 69.

At one extreme of the uncertainty relation between amplitude and temporality, the ideal sine wave would represent a timeless series. Squeezed through an impossibly narrow filter, the signal would be cleansed of both temporality and all possible noise. Widening the bandwidth of this hypothetical, ideal filter, however, would allow a larger frequency spectrum to seep through. In accordance with the uncertainty principle, this would shorten the filter's response time and thus its delay: the temporal factor t would cease to be an idealized $t = \infty$ and revert back into a finite, physical timeframe. Widening the bandwidth and shortening response and delay still further, one ultimately arrives at the other extreme of the uncertainty relation. Here we find another analytical idealization, which precisely inverts the ideal series. With a symbolic delay time of 0, the timeframe of this ideal transmission would be reduced to an infinitesimally short moment. This is a Dirac impulse or delta function, named after British physicist Paul Dirac or the sign used to represent the function, the · . A Dirac impulse represents the radical instantaneity of something that happens in less than a flash, the ultimate transient phenomenon: the ideal event.

The Dirac delta is a peculiar function with an infinitesimally short timeframe and—when t is exactly zero—an infinite amount of energy.[40] In terms of our understanding of the uncertainty principle, the function implies a filter with an instantaneous response and a delay time of 0. Given that a filter's response time affects its precision (the longer the response time, the more precise the filtering), a hypothetical filter with a response time of 0 would filter nothing out: all frequencies would pass through it unfiltered. In consequence, a Dirac impulse's frequency spectrum is infinite, as illustrated by the upward-facing arrow in Figure 3.1. Here, an infinite number of frequencies occur at one, infinitesimally short moment. Accordingly, the Dirac impulse inverts the sine wave exactly: whereas the latter represents one frequency repeating infinitely, the former contains all frequencies in an infinitesimally short time.

Like the sine wave, the delta function is a symbolic limit that does not directly represent anything in the physical world but can only be approximated. The more closely a real signal approximates the infinitesimally short spike of a Dirac

[40] Although Dirac formally defined the delta function and suggested its standard notation in his *Principles of Quantum Mechanics* in 1930, the concept itself already appears in the step function proposed by the nineteenth-century mathematician and physicist Oliver Heaviside. Graham Farmelo, *The Strangest Man: The Hidden Life of Paul Dirac, Mystic of the Atom* (New York: Basic Books, 2009), 113. As mathematician Jesper Lützen notes, the delta function ultimately follows from results with which Fourier himself was "confronted" in the early nineteenth century. Jesper Lützen, "Between Rigor and Applications: Developments in the Concept of Function in Mathematical Analysis," in *The Modern Physical and Mathematical Sciences*, ed. Mary Jo Nye, Volume 5 (Cambridge: Cambridge University Press, 2002), 479–480. Accordingly, both the Dirac delta and sine wave originate in the same discursive context.

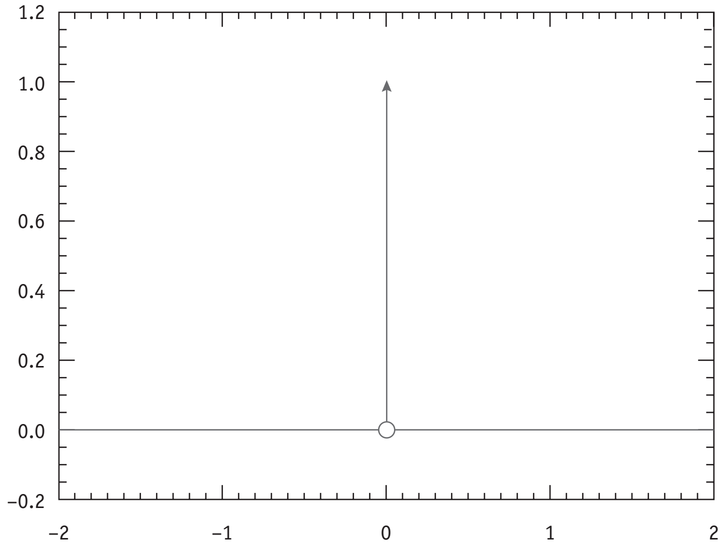

Figure 3.1 The Dirac Impulse or Delta Function. A graphical representation of the Dirac-delta: the upward direction of the arrow, the width of which should ideally be infinitesimally small, indicates that its frequency spectrum is infinite. This figure illustrates why the Dirac impulse is also referred to as the "needle function." (Courtesy of Omegatron, "Dirac distribution PDF," adapted by Qef, *Wikimedia*, accessed October 23, 2015 commons.wikimedia.org/wiki/File:Dirac_distribution_PDF.svg).

delta, the more frequencies it contains.[41] Kittler's metaphor of a bolt of lightning is well chosen: its enormous energy discharge contains a frequency spectrum so large as to approximate a Dirac impulse. Musically, as well, one can compare a bow gently stroking a violin string with a short smash on a snare drum. The bow's long gesture slowly sets the string's vibrations in motion, producing a semiperiodic and primarily harmonic frequency spectrum. These vibrations tend toward an ideal sine wave (but will nonetheless fall short, due to the vibrating string's rich harmonic overtones). In the case of the snare drum, the sudden noisy blow produces a broad and complex nonperiodic frequency spectrum. Full of inharmonic partials, this sound approximates a Dirac impulse.[42]

[41] Reportedly, Dirac's training as an engineer led him to "tolerate" such approximations: "the pure mathematician who wants to set up all of his work with absolute accuracy," he said, "is not likely to get very far in physics." Dirac in Paul Nahin, *Dr. Euler's Fabulous Formula Cures Many Mathematical Ills* (Princeton: Princeton University Press, 2006), 192.

[42] In communication engineering, a physical, real-time approximation of a Dirac impulse function (in the form of a very—but not infinitesimally—short blow) is used to test a system's frequency response. By feeding the impulse to its input and measuring which frequencies resonate within the system itself, it is possible to discover which elements, materials, or room acoustics affect (shape, change, or dampen) the

Just as the sine wave can be interpreted as the product of an infinitely accurate spectral filter, which perfectly isolates an unambiguous signal amid an infinite sea of frequencies, so a Dirac impulse can be interpreted as the product of an ideal temporal filter, which extracts one infinitesimally short instant from the flow of time. The infinite timeframe of the ideal sine wave turns constantly changing sound waves into an endlessly repetitive rhythmical order, effectively bringing time to a standstill. A Dirac impulse, by contrast, does not "last" any amount of time, for reducing time to its absolute zero value means that it can no longer be understood in terms of duration. Time, here, is a pure, point-like "present": an impossibly exact moment or absolute now, without any connection to past or future. Diametrically opposed to the Fourier domain's complete stasis, the upward-pointing needle of the Dirac impulse represents a singular event: pure transience.

Only a device capable of perfect accuracy and unlimited resolution would be able to capture and analyze the spectrum of such an event at the very moment it occurs. That device would instantaneously register, to use Kittler's vocabulary, both that it took place and what it was. Infinitely perfect analysis, then, requires a combination of the infinitesimal temporal precision (absolute singularity) of Dirac impulses and infinite spectral clarity (infinite periodicity) of sine waves. The pure event can never be analyzed completely, however, because it contains an infinite number of frequencies. The pure series, conversely, can never be fully processed, for that would take an infinite amount of time. According to the uncertainty principle, whatever we might gain in one domain, we lose in the other. Hence, although the Fourier transform might seem to unravel and demystify the complexity of noise by posing an ordered series of sine waves, this comes at the cost of doing away with all instantaneity and transience. At the other extreme of this uncertainty principle, the infinitesimal window of the Dirac impulse is nothing but transience. By turning our attention away from the purity and clarity of the Fourier domain and toward the temporal filter of the delta function, then, we can explore

output. Here, the "thatness" of an approximate Dirac delta is used to measure the characteristic "whatness" of an (electro)-acoustic system, and test how these characteristics might affect a signal fed to its input. A similar principle is central to pulse code modulation (PCM), the most common method for sound digitization, which I discussed in chapter 1. To determine the amplitude values of each sample, an analog-to-digital converter runs a series of pulses, each of which, writes Kadis, "approximates an impulse, an infinitely narrow pulse." Kadis, *Science*, 149. Before they are modulated, Rumsey and McCormick explain, "all these pulses have the same amplitude (height), but after modulation the amplitude of the pulses is modified according to the instantaneous amplitude of the audio signal at that point in time." Rumsey and McCormick, *Sound and Recording*, 211. Hence, modulating a regular series—in most cases 44,100 per second—of approximate Dirac impulses with an analog audio waveform can scale impulses so as to match the waveform's amplitude at that particular point in time. This results in a time-limited sample of a band-limited signal, which is the basis for all digital audio. It also shows that the ideal timeframe of a digital sample would be the infinitesimally short timeframe of a Dirac impulse.

in more depth the randomness and transience that escapes the spectral filters of Fourier analysis.

The Domain of Technical Filters

Constrained by the laws of the uncertainty principle that I have been discussing, physical signals must strike a balance between precision in time and precision in amplitude. In an essay first published in 1967, Norbert Wiener recounts how he used a musical example to explain the physical consequences of this balancing act at a talk given in Göttingen in 1925.[43] Every waveform, he writes, occurs within a finite (but not infinitesimal) timeframe, which lasts long enough to allow the waveform to complete at least one full cycle. In the case of complex harmonic waveforms (including most musical sounds), a hearer can generally identify the characteristics of a specific frequency spectrum once the fundamental frequency (which determines the fundamental pitch) completes at least one cycle. This, Wiener explains, means that "if you take a note oscillating at a rate of sixteen times a second and continue it only for one twentieth of a second, what you will get is essentially a single push of air without any marked or even noticeable periodic character."[44] When the waveform's fundamental frequency is cut short, the physical signal "will not sound to the ear like a note," but will rather resemble a short, transient blow, impulse, or noise.[45]

This is the uncertainty principle at work. At a certain limit, tending toward but never reaching the infinitesimally short timeframe of a Dirac impulse, it becomes impossible to shorten a sound without losing its identifiable frequency spectrum in the process. Beyond this threshold, the analytical clarity of spectral analysis gives way to the instantaneity of Dirac impulses: clearly definable sine waves disappear into fuzzy, undefined spectra and all that remains is a transient blow or noise. This is why, writes Wiener, "you can't play a jig on the lowest register of the organ": the jig's tempo is faster than the time it takes for the lowest frequencies to finish one cycle.[46] Besides demarcating the musical limits of organ performances, these fuzzy, nonperiodic transients are an indispensable aspect of every sound. Physical sounds do not exist at

[43] Wiener, "Spatio-Temporal Continuity," 539–546.
[44] Wiener, "Spatio-Temporal Continuity," 545.
[45] Wiener, "Spatio-Temporal Continuity," 545.
[46] Wiener, "Spatio-Temporal Continuity," 545. Kittler writes in "Lightning and Series—Event and Thunder": "Before a deep organ tone can turn into an event, many high trebles have already been recognized." Although he does not credit Wiener, it is likely that Kittler drew the example from his paper. Kittler, "Lightning," 71.

either one of the limiting cases of the uncertainty relation. Between the two extremes of sine waves and Dirac impulses, every signal has a beginning, duration, and end. Even an almost entirely periodic signal does not continue forever: at some point—even if only with the final collapse of the universe—it will stop.

Absolute purity requires a timeframe that stretches infinitely into the past and the future. No finite filtering operation, therefore, will attain perfection and no real signal is entirely pure. Transience negates the Fourier domain's infinite periodicity, for the inevitable starts and stops of every signal cause, Wiener puts it, "an alteration of its frequency composition which may be small, but which is very real."[47] This means that the beginning and end (in acoustic terms, the "attack" and "decay") of every sound add elements of nonperiodic transience. These random alterations give a sound its unique timbral quality. Whereas periodic frequencies are largely responsible for determining pitch and overall harmonic composition, these nonperiodic alterations determine the specific tone color (and also, in the case of speech, specific vowel color and consonant shape). Composer Henry Cowell calls them the "noise element in the very tone itself."[48]

These sonic traces of attacks and decays mark the difference between the plane of the ideal filter and physical sounds produced in the domain of technical filters. The symbolic gesture of a clean cut, administered by the ideal filter, separates the former from the latter sphere, seamlessly removing the singular event from its natural flow and turning it into an infinite series.[49] This cut

[47] Wiener, "Spatio-Temporal Continuity," 544–545.

[48] Henry Cowell, "The Joys of Noise," in *Audio Culture: Readings in Modern Music*, eds. by Christoph Cox and Daniel Warner (New York: Continuum Group, 2004), 22–24. "Consider the sound of a violin," Cowell writes, "part of the vibrations producing the sound are periodic, as can be shown by a harmonic analyzer. But others are not—they do not constantly reform the same pattern, and consequently must be considered noise. In varying proportions all other instruments yield similar combinations."

[49] In his discussion of Kantian aesthetics in *The Truth in Painting*, Jacques Derrida writes about the "*sans* of the pure cut" [*Le 'sans' de la coupure pure*]: "So it is the *without* that counts for beauty; neither the finality nor the end, neither the lacking goal nor the lack of a goal but the edging in *sans* of the pure cut, the *sans* of the finality *sans*-end." Conceptually, this paradox of the finitude inherent to an actual cut and the ideal infinity of a pure cut, which leaves no traces of its cutting, bears similarities to my idea of the clean cut. Jacques Derrida, *The Truth in Painting*, trans. Geoff Bennington and Ian McLeod (Chicago: University of Chicago Press, 1987), 89, emphasis in original. In a different context, in *Meeting the Universe Halfway*, Karen Barad describes the problem of separating subject and object, observer and observed in quantum mechanics (which, given the importance of the uncertainty principle, is not that far removed from the present discussion): "So the question of what constitutes the object of measurement," she writes, "is not fixed: as Bohr says, there is no inherently determinate Cartesian cut. [. . .] What constitutes the object of observation and what constitutes the agencies of observation are determinable only on the condition that the measurement apparatus is specified. The apparatus enacts a cut delineating the object from the agencies of observation. Clearly, then, as we have noted, observations do not refer to properties of observation-independent objects (since they don't preexist as such)." In this example, as with my concept of the clean cut, the cut itself is (at least partly) constitutive for establishing a more or less unambiguous object of observation and analysis. Karen Barad, *Meeting the Universe Halfway: Quantum Physics and the Entanglement of Matter and Meaning* (Durham: Duke University Press, 2007), 114.

transforms temporal events into infinitely oscillating frequencies, doing away with all temporal or spectral randomness, and thus with all possible noise too. Indeed, the literal and figurative figure of the cut appears throughout discourses on technical media and sound recording. The common expression to "cut" a record describes the way that grooves used to be cut into the recording material (wax, acetate, vinylite). The phrase also has a more metaphorical resonance. Oliver Read, for example, recommends recording styli "that produce quiet, clean cuts."[50] The cleanliness of the cut, here, refers not only to the technological procedure of cutting grooves, but also to the sound quality of the recording itself, which should be "cut" with as few acoustic traces as possible.[51] The common expression of "cutting" a track extends this double-sided trope to magnetic tape recording, referring to both the literal "cutting" of tape and metaphorical "cutting" of a piece of music from its sonic flow. Such cuts craft a more or less clearly delineated musical object, separating one song or track from another. As such, the symbolic gesture of the clean cut is fundamental to the myth of perfect fidelity and conceptual logic of noise reduction.

In contrast to the clean cuts made by an ideal filter, technical filtering operations apply physical cuts that must strike a compromise between the spectral domain and time domain, that is, between sine waves and Dirac impulses. Set apart from symbolic representations and the ideal filter, the operations of physical filters process real-time signals that extend in space and change over time. In the previous chapter, my analysis of dual-ended noise reduction and dithering showed how the conceptual logic of noise reduction presupposes a clear, unambiguous definition of what constitutes noise and what signal. Such a distinction amounts to a clean cut, executed by a perfect symbolic noise filter, separating everything that you want from everything that you do not. In the domain of technical filters, the noisy traces of physical filtering operations, applied by technical sound media, signify the impossibility of clean cuts.

As soon as a signal starts, it introduces nonperiodic oscillations and transient events, adding randomness, unpredictability, and spectral complexity. In sharp contrast to the infinite repetition of sine waves, such transience

[50] Read, *Recording*, 46.

[51] Of course, magnetic tape also enabled cutting and pasting as an aesthetic tool, a means of both separating and ordering sonic material and of disruption. William Burroughs's famous "cut-ups" in the 1960s, "where he," as N. Katherine Hayles describes, "physically cuts up previously written narratives and arbitrarily splices them together," use a technique that was and still is heavily employed in both avant-garde and pop musical practices. These practices, in turn, take their cue from modernist cut-up and cut-and-paste practices developed in early twentieth-century avant-garde movements. N. Katherine Hayles, "Voices Out of Bodies, Bodies Out of Voices: Audiotape and the Production of Subjectivity," in *Sound States: Innovative Poetics and Acoustical Technologies*, ed. Adalaide Morris (London: University of North Carolina Press, 1997): 88.

makes each moment sonically different from the next. Each link in the chain from sender to receiver (each passageway or gate) filters the sound in specific ways, adding transient noises to the signal. Every technologically processed sound contains traces of every incidental or technical thing—air, copper, or glass fiber—that it has encountered in the acoustical, electro-acoustical, electronic, or digital domains and bears marks of the specific circumstances through which it has unfolded: humidity, air pressure, altitude, etc. etc. Wiener describes these noises as "small" but "very real" alterations of the frequency composition; von Neumann called them the "small extra" added to the output. At every stage of sound reproduction, they change the sound.

In 1953, while visiting a radio studio in Brussels, Belgian composer Karel Goeyvaerts, learned of a machine that could electronically generate sine waves. This machine, he realized, might offer an opportunity for creating a type of music about which he had been speculating for quite some time. Ever since they had met at the famous Summer Courses for New Music in Darmstadt two years earlier, Goeyvaerts and his German friend and colleague Karlheinz Stockhausen had been exchanging letters in which they discussed, among many other things, the musical, aesthetic, and formal foundations of total serialism. The two composers had been developing this compositional principle alongside Frenchman Pierre Boulez and Italian Luigi Nono. Expanding on Arnold Schönberg's prewar twelve-tone system, total serialism did more than just make the twelve tones of the Western diatonic scale equally important. Beyond this, it strived to rationalize each musical parameter: duration, volume, meter, and the spectral composition of sounds. As their correspondence attests, during the early 1950s, Goeyvaerts and Stockhausen considered the sonic purity of sine waves key to this quest for absolute control over the compositional material. Indeed, for Goeyvaerts, sine waves were nothing less than the "almighty basic material governing every sound phenomenon."[52]

Even before discovering the sine wave generator, Goeyvaerts had already written *Nr.4 met Dode Tonen* (*N°.4 with Dead Tones*) (1952). Although he provided technical and musical instructions for this electronic composition, it was not realized sonically until several decades later.[53] The "dead tones" mentioned in the title were to be pieced together from what Goeyvaerts at that point called "sound atoms."[54] Although they could have a complex frequency

[52] Karel Goeyvaerts, *Selbstlose Musik: Texte, Briefe, Gespräche*, ed. Mark Delaere (Cologne: Musik Texte, 2010), 166.
[53] Karel Goeyvaerts, *Compositie Nr. 4*, track 4 on *The Serial Works* [#1–7], Megadisc Classics, 1998, compact disc.
[54] Goeyvaerts in Herman Sabbe, "A Paradigm of 'Absolute Music': Goeyvaerts's N°.4 as 'Numerus Sonorus,'" *Revue Belge De Musicologie/Belgisch Tijdschrift voor Muziekwetenschap* 59 (2005): 243.

composition, these tones should have no unpredictable "inner life," as Richard Toop puts it.[55] That is to say, they had to "be identical at any moment in time, and therefore detached from time itself."[56] By using the most static sounds, which do not suffer from the irregularities introduced by attacks and decays, Goeyvaerts wanted to minimize sonic transience and approximate the characterless immortality of ideal sine waves.

This desire to technologically "lift the sense of time," as Herman Sabbe has written, became even more pronounced with *Nr.5 met Zuivere Tone*n (*N°.5 with Pure Tones*). Goeyvaerts conceived the piece after discovering the sine wave generator that enabled its actual production.[57] By building the spectral composition of the eponymous "pure tones" from the ground up using electronically produced sine waves, *N°.5* took the rationalist logic of total serialism to its extreme. The pure tones were to be constructed out of individual sine waves following the same rules that organized the other parameters. Furthermore, like *N°.4*, the piece is entirely symmetrical or palindromic (turning around at its half point and closing in on itself at the end). Accordingly, *N°.5* is an almost algorithmic procedure, intended to be sonically identical to itself at all times and with each playback.[58]

Ultimately, Goeyvaerts was dissatisfied with the sound of *N°.5*. As he recalled in 1994, there was "a lot of crackle on the tape," and far from achieving a blend of "more bright or more muffled" sounds, "one could clearly hear the different component tones" of each sound.[59] Instead of achieving sonic purity and clarity, Goeyvaerts realized that "absolute certainty lay outside my grasp."[60] It is therefore not in fulfilling, but rather in performing the unrealizability of Goeyvaerts's ideals that these pieces encapsulate the fundamental role of the noise resonance of sound media. They show how the material agency of the medium itself defines the ways in which technologically produced sounds make musical sense. While attempting to transcend the material basis of sound production and make music with an ideal filter, Goeyvaerts was confronted with the unruliness of technical filters. Purity, he found, lies forever out of reach. In accordance with the mathematical principles of Fourier analysis, Goeyvaerts's dead and pure tones aspire to timeless

[55] Richard Toop. "Stockhausen and the Sine-Wave: The Story of an Ambiguous Relationship," in *The Musical Quarterly 65*, no. 3 (1979): 386.
[56] Toop, "Stockhausen," 386.
[57] Herman Sabbe, "Goeyvaerts and the Beginnings of 'Punctual' Serialism and Electronic Music," *Revue Belge De Musicologie/Belgisch Tijdschrift voor Muziekwetenschap* 48 (1994): 76. Karel Goeyvaerts, *Compositie Nr.5*, track 5 on *The Serial Works* [# 1–7], Megadisc Classics, 1998, compact disc.
[58] Sabbe, "Paradigm," 243.
[59] Karel Goeyvaerts, "Paris-Darmstadt 1947–1956. Excerpt from the Autobiographical Portrait," *Revue Belge de Musicologie/Belgisch Tijdschrift voor Muziekwetenschap* 48 (1994): 51.
[60] Goeyvaerts, "Paris-Darmstadt," 51.

clarity. However, this transcendental desire ran up against the physical fact that sounds simply cannot be entirely static and frozen in time. The channel itself always introduces Wiener's "small but very real," alterations and Von Neumann "small extra"—noises, distortions, and transient alterations to the frequency spectrum.

Along every physical channel, these alterations cling to the sound signal and cause the output to differ ever so slightly from the input. In terms of both information theory and thermodynamics, such difference increases entropy. Indeed, in a 1949 letter to Gödel, Einstein wrote, "the sending of a signal is, in the sense of thermodynamics, an irreversible process, a process which is connected with the growth of entropy."[61] The arrow of time flies from the past to the future and it cannot, as Goeyvaerts wished, stand still. The ways in which material transmission channels physically change signals therefore constitute temporal traces of the transmission itself. Accordingly, these changes not only signify the difference between input and output, or original and copy. More importantly, they signify a difference in time. In the next chapter, I show how this "time critical" character of transients constitutes a crucial aspect of the noise resonance of sound media.[62]

[61] Einstein in Kurt Gödel, "Static Interpretation of Space-Time with Einstein's Comment on It," in *The Concepts of Space and Time: Their Structure and Their Development*, ed. Milič Čapek (Dordrecht: D. Reidel Publishing Company, 1976), 459.

[62] Wolfgang Ernst, *Gleichursprünglichkeit: Zeitwesen und Zeitgegebenheit von Medien* (Berlin: Kadmos, 2012), 39–43.