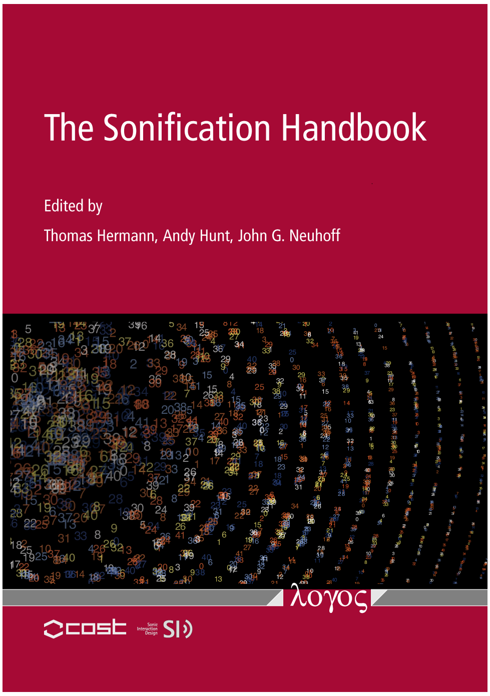

The Sonification Handbook

Edited by
Thomas Hermann, Andy Hunt, John G. Neuhoff

Logos Publishing House, Berlin, Germany
ISBN 978-3-8325-2819-5
2011, 586 pages

Online: http://sonification.de/handbook
Order: http://www.logos-verlag.com

Reference:
Hermann, T., Hunt, A., Neuhoff, J. G., editors (2011). *The Sonification Handbook*. Logos Publishing House, Berlin, Germany.

Chapter 12

Audification

Florian Dombois and Gerhard Eckel

This chapter introduces the sonification technique of Audification. Some definitions are given, an overview of the history of Audification in the Sciences and in the Arts and an extensive description of how to make use of this sonification technique.

Reference:
Dombois, F. and Eckel, G. (2011). Audification. In Hermann, T., Hunt, A., Neuhoff, J. G., editors, The Sonification Handbook, chapter 12, pages 301–324. Logos Publishing House, Berlin, Germany.

Media examples: http://sonification.de/handbook/chapters/chapter12

Chapter 12

Audification

Florian Dombois and Gerhard Eckel

12.1 Introduction

Music is ephemeral. It intrudes into reality for a moment, but escapes it in the next. Music has a form, which is expressed over time, but cannot be touched. Evidently, music is difficult to archive, and two main techniques have challenged its transience: (i) the *score*, as a code for instructing instrumentalists or other sound generators for later re-enactment, and (ii) the *recording*, as the acoustic complement to photography, which registers the sound wave at a specific point of listening. Both techniques have their advantages and disadvantages since they cannot exactly repeat the original, but they both open a particular perspective on the original sound and disclose aspects perhaps otherwise not to be heard. The first approach stresses, for example, more the tones and their symbolic value, whereas the second traces the exact physical wave in an analog manner. In sonification, one can find these two perspectives too: (i) the technique of *parameter mapping* (see chapter 15), and (ii) the technique of *audification*. This chapter concentrates on the second.

Gregory Kramer defines in his book *Auditory Display*: "The direct playback of data samples I refer to as 'audification"' [32, p. xxvii]. And as a later update of this definition: "Audification is the direct translation of a data waveform into sound." [60, p. 152]. The series of data might not even belong to the sound domain. A common way of displaying this visually would be a Cartesian graph. If the visualized data have a wave-like shape, e.g., an EEG signal, audification would mean to attribute their values to air pressure, and transferring the result to a loudspeaker, whereby the data then become audible. The aim behind this media shift, as in all sonification techniques, is that the other mode of representation discloses or makes emerge aspects of the data that might not have been discovered before. It is a direct alternative approach to visualization, since all abstract data series can be either visualized or sonified. So one might define: *Audification is a technique of making sense of data by interpreting any kind of one-dimensional signal (or of a two-dimensional signal-like data set) as amplitude over time and playing it back on a loudspeaker for the purpose of listening*. And since all data end up in a loudspeaker, audification is essentially a continuous, non-digital interpretation of data sets.

In audification one can distinguish between different types of data that result in different types of sounds. Often, sound recordings themselves have already been named "audification" if they have been shifted in pitch. Therefore, we want to include with our definition above all data sets that can be listened to, i.e. also all sound recordings themselves. We see four groups of data (see Fig. 12.1, withdrawing more and more from the audio-context: (i) sound recording data, (ii) general acoustical data, (iii) physical data, and (iv) abstract data.

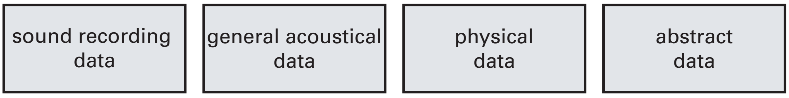

Figure 12.1: Classification of data for audification

(i) Sound Recording Data: The first group of data, to which audification can be applied, are sound recordings, which are today normally sampled digitally as series of numbers. Taking these time series one could say that every CD-Player has an audification module, which is the Digital-to-Analog (DA)-converter transforming the series of data points into a continuous sound signal. Now, usually there is little special from the viewpoint of sonification about listening to sound recordings themselves. This becomes different when sound recordings are amplified, thereby revealing unheard aspects in the recordings. And it becomes even more interesting when the recordings are time-compressed or -stretched. For example, ultrasonic signals such as bat calls are inaudible to the human ear unless they are transposed (sound examples **S12.1**, **S12.2**). The change of playback speed is then certainly more than a gimmick, and audification can function in this context as an acoustic micro- or telescope.

(ii) General Acoustical Data: All kinds of measurements in elastomechanics, which follow the same physical laws as an acoustic wave, constitute a major area of interest for audification. In particular, vibrational data of mechanical waves are easily accessible by listening to their audification. From applying our ears to a railroad rail, a mast or a human belly, from using sounding-boards, stethoscopes or sonar, we are familiar with interpreting mechanical waves acoustically. And, even though they are always a combination of compressional and transversal waves, the character of mechanical waves is usually preserved when being reduced to a one-dimensional audio signal. Changing the playback speed also usually proves to be of minor influence on the plausibility of the resulting sound. This is especially evident in Auditory Seismology, where seismograms are often audified with acceleration factors of 2,000 and more (sound example **S12.3**) [15].

(iii) Physical Data: There are measurements of other physical processes outside the mechanical domain that can be audified too. But these data, for example electromagnetic waves, usually lack acoustic familiarity with our daily hearing. The different velocities of wave-propagation or of the dimensions of refraction and reflection effects etc. result in a new soundscape unfamiliar to human experience. Therefore, one has to be careful with

interpretation; e.g., EEG data (sound example **S12.4**) of several electrodes around a head cannot simply be compared to a similar arrangement of microphones within a room.

(iv) Abstract Data: The lack of acoustic familiarity may worsen when using abstract data for audification which do not stem from a physical system. Examples of this non-physical data might be stock market data, or when listening to a fax-machine (sound example **S12.5**) or a computer-modem at the telephone (sound example **S12.6**). Not all wave-like shapes in abstract data conform to the wave equation, therefore interpreting those audified signals usually takes more time for the user to become habituated. Nevertheless, non-acoustic and abstract data can easily be audified when they are arranged as a time series.[1]

Audification is the simplest technique of sonification and is therefore often used as a first approach to a new area of investigation, but then mostly neglected in the further development of sonification projects. This chapter hopes to show that there are many reasons not to undervalue the potential of audification, especially when adding various acoustic conditioning techniques as described in Section 12.3. It also hopes to widen the scope for development towards more interactivity in the use of audification parameters.[2] The remainder of the chapter is organized as follows:

Section 12.2 gives a brief introduction to the history of audification from the 19th century until the first ICAD in 1992. Here, the focus is not only on the history of science, but also some examples from the history of music and art. Section 12.3 is dedicated to the technical side of audification and tries to unfold many possibilities for optimizing the acoustic result of audification. Audification can seem to be a simple reformatting procedure only at a first glance. Instead, it can be much more sophisticated when extended to the art of sound processing. Section 12.4 summarizes the areas in which audification is used today, especially referring to the ICAD and its papers. Section 12.5 gives some rules of thumb, how and when to use, and what to expect from the audification of a data set. Finally, Section 12.6 outlines the suggested next steps that need to be taken in order to advance the application of audification.

12.2 Brief Historical Overview (before ICAD, 1800-1991)

Three inventions from the 19th century are of great importance to the history of sonification: (i) the *telephone,* invented by Bell in 1876, (ii) the *phonograph,* invented by Edison in 1877, and (iii) *radiotelegraphy,* developed by Marconi in 1895. The transformation of sound waves into electric signals, and vice versa, started here, and the development of the loudspeaker began as a side product of the telephone. These tools for registering and displaying sound gave rise not only to a new era of listening,[3] but also to the research field of audification. If we take the "Time Axis Manipulation" of *sound recording data* as the simplest form of intentional audification, we find Edison demonstrating this technique already in 1878 in New York [24, p. 27f.] and in the 1890s the Columbia Phonograph Company suggested reversing the direction of playback as an inspiration for new melodies. [24, p. 52].

As well as reproduction, mediation is an important part of audification. The whole idea of

[1] This classification of data in four groups could be developed further and one could discern, for example, in each of the four between continuous and discrete or digital datasets (cf. Section 12.3), etc.

[2] Cf. [29, p. 21] and [28, p. 6]

[3] Cf. [58] for a profound investigation of how much these inventions changed our relation to hearing in general.

data-driven sound, which is the key concept of sonification, could only be made possible by the introduction of a medium, which, as electricity or an engraved curve, also makes it possible for all forms of data to be displayed in the sound domain. By extending the human auditory sense with technology, the process of listening to data can be thought of as involving data, conversion, display and perception.

12.2.1 Listening to the measurement in science

In science, visualization has played the dominant role for centuries, whereas sound and listening to natural phenomena has always been under suspicion. Nevertheless, there are a few exceptions. One of the very early uses of scientific audification of *general acoustical data* (cf. Section 12.1), even before the electrical age, was that made by the French doctor R. T. H. Laënnec who, in 1819, invented the stethoscope [35]. This auditory device, which is still in use, had a great career as an instrument for medical diagnosis, especially after being redesigned by Georg Philip Camman, and is one of the few important examples of an accepted scientific device using audio[4] (sound example **S12.7**). Auenbrugger later added "interactivity" to it by introducing percussion and gave listening into the human body another boom of success [2]. This has not stopped, and we find similar instruments handling mechanical waves even in today's plumbers' equipment to track leaking conduits ("Hördosen").

In audifying *physical data*, as introduced above, the earliest examples date back as early as 1878, when a series of papers was published about connecting muscle cells with the newly invented telephone.[5] These groundbreaking publications have not been considered by the ICAD community, so far as we know, but, for example, J. Bernstein and C. Schönlein, in their paper of 1881[8], describe nothing less than how they studied the reaction frequencies of muscle cells and the transmitting qualities of the cells as what they call "muscle telephone" [8] (p. 27) by listening to its audification. The first audification of nerve currents was published a little later by Nikolai Evgenievic Wedenskii, in 1883, also using the loudspeaker of a telephone as an audio display of physiological processes [61]. Later, in 1934, a few years after Hans Berger's famous publication of EEG waves, E.D. Adrian and B.H.C. Matthews proved his experiments also using audification [1]. Another successful example of audifying non-acoustic data is the Geiger counter, which was invented by Hans Geiger in 1908 and developed further in 1928, resulting in the Geiger-Müller tube, which is still used today (sound example **S12.8**).

Almost at the same time is found what is probably the first use of Time Axis Manipulation in the scientific context: US scientists applied new methods of sound processing to the echo-locating sounds of bats and, in 1924, released a record with transposed bat recordings now audible for the human ear [31, p. 152]. Also, the technique of sonar (SOund Navigation And Ranging) dates from this time, even though the idea apparently can already be found in Leonardo da Vinci's manuscripts [33, p. 29]. It was developed during World War I in Great Britain to track submarines. The use of the Vocoder as a sound encoding device seems to be worth mentioning here. In the SIGSALY system for secure voice transmission

[4] There are several training websites for doctors to learn auscultation, e.g. http://www.wilkes.med.ucla.edu (accessed Jan 30 2011)

[5] The first paper, we found, is [26]. For more material cf. [17] and [18]. There is also an ongoing research project, "History of Sonification" by Andi Schoon and Axel Volmar, who have found about 30 more articles in that early era. First publications are [57], [59] and [56], but please watch out for their coming publications.

developed in 1942-1943, the Vocoder would encrypt the speech signal by another sound signal, for example with a noise recording. The transmitted signal would then sound like random noise, but at the receiving station the same noise sequence would be used to decode the speech signal, thus making the voice recognizable again [31, p. 78]. Depending on the encrypting signal the encrypted voice can become of interesting fashion, an effect that has been used in many science-fiction movies (sound example **S12.9**) and pop songs (sound example **S12.10**).

We have already mentioned that inventions of new technologies and media have a major influence on scientific development. One important improvement for audification research happened when audio tape was developed by Fritz Pfleumer in 1928, on paper, and later in 1935 by BASF, on plastic tape. This new material was soon used as a common data storage medium, and already Alan Turing seemed to have thought of magnetic audio tape as storing material for the upcoming computer. Using a recording and a playback head, allowing backward and forward play, magnetic tape was the start for "every thinkable data manipulation".[6] This new sound recording material transformed the temporal phenomenon of sound, even more explicitly than the wax roll, into a trace on a substrate – time is linearly represented in space. By manipulating the substrate (e.g., cutting and splicing a tape), sound can be transformed out-of-time, and Time Axis Manipulation becomes feasible through changing the reading speed of the substrate. This might explain the great success of audio tape after World War II and its intense use in all kinds of application areas. Among others, in the 1950s, seismologists started to write seismological recordings directly on magnetic tape in an audio format and several researchers listened to their data first for entertainment, we have been told, but then figured out that this method is especially valuable for detecting signals of seismic events in noisy records. The first scientific paper on audifying seismic recordings is "Seismometer Sounds" by S. D. Speeth, in 1961, where he describes testing audification for signal discrimination between natural quakes and atomic explosions (cf. [55]; see also [21]).

12.2.2 Making the inaudible audible: Audification in the arts

Interestingly enough, audification not only has its roots in science but also very much in the arts. The first text that is known on an audification of *abstract data* can be found with Rainer Maria Rilke, a German poet, dating back to 1919. "Ur-Geräusch" (Primal Sound) [52] is a short essay reflecting on the form of the coronal suture of a skull, imagining the shape translated into a sound. Another important text was written by László Moholy-Nagy, in discussion with Piet Mondrian in 1923, where he wants to compose New Music by etching sound curves directly on the record [43], which is nothing less than an audification of graphical lines, long before the computer was invented. Ideas similar to that of etching a disc were brought up by introducing soundtracks to film in the 1920s. Oskar Fischinger, in 1932, in reaction to the famous "Farbe-Ton-Kongreß" in Hamburg[7], started to investigate painting ornaments directly on the soundtrack of a film (see Figure 12.2).[8] This technique resulted in new synthetic sounds, very much like those from electric synthesizers, and produced a huge

[6] [31, p.165] ("jede erdenkliche Manipulation an Daten")

[7] In 1927 and 1930 the first two conferences on the relation between color and tone were held in Hamburg, Germany, that had a major impact on the development of synaesthetic art.

[8] First published in *Deutsche Allgemeine Zeitung* 8.7.1932; cf. [44, pp. 42-44]

press reaction in Europe, the US, and even in Japan. Nevertheless, Fischinger unfortunately could not get a grant for further research and released only a few recordings, which at least were well-received by John Cage and Edgard Varèse.

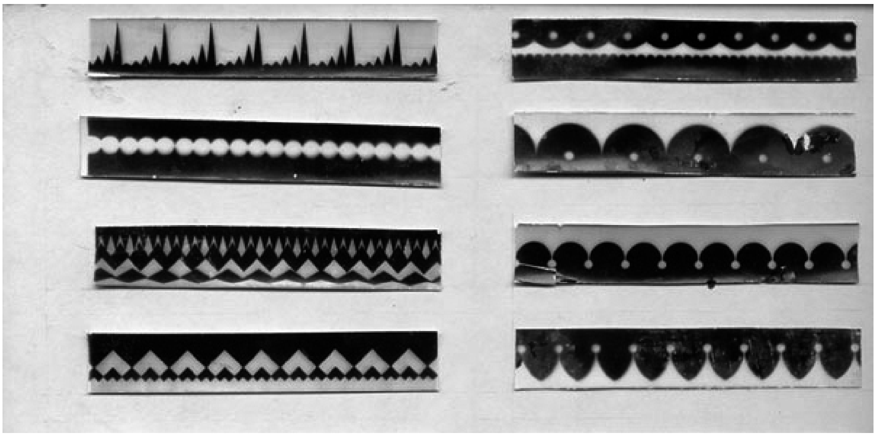

Figure 12.2: Detail from Oskar Fischinger's studies of sounding ornaments.

Another genealogic line follows composers of New Music using new technology for their compositions. For example in 1922 Darius Milhaud began to experiment with "vocal transformation by phonograph speed change" [54, p. 68] in Paris and continued it over the next 5 years. The Twenties also brought a wealth of early electronic instruments — including the Theremin (sound example **S12.11**), the Ondes Martenot (sound example **S12.12**) and the Trautonium (sound example **S12.13**), which were designed to reproduce microtonal sounds. The historical beginnings of electronic music can also be interpreted as a history of audification, because all electronic instruments use electric processes audified on a loudspeaker. This chapter does not describe further this rich story of further inventions of data processing with acoustic results (i.e., the history of electronic music), since it moves too far away from the core of audification and must defer here to the existing literature.

But at least two more developments seem worth mentioning here:

(i) In 1948 Pierre Schaeffer developed his idea of "musique concrète" which was the furious starting point for composing with recorded material and sound samples in music. Naturally, all forms of manipulation and conditioning of the acoustic material were developed subsequently (Time Axis Manipulation, all sorts of transpositions, reverse playing, filtering etc.), first mainly with audio tape and record players, and later with the computer.

(ii) Another influential development was the artists' interest in the unheard sound cosmos. For example Jean Cocteau's reaction to the discoveries of ultrasounds was an enthusiastic conjuration of yelling fish filling up the sea with noise.[9] And many projects, like the LP record "BAT" of Wolfgang Müller (1989) (sound example **S12.14**) with down-pitched ultrasounds,

[9] Cf. [10, p. 36f.]: "Die Welt des Tons ist durch die noch unbekannte Welt des Ultraschalls bereichert worden. Wir werden erfahren, daß die Fische schreien, daß die Meere von Lärm erfüllt sind, und wir werden wissen, daß die Leere bevölkert ist von realistischen Geistern, in deren Augen wir ebenfalls Geister sind."

received their attention in the art world because they displayed another world beside ours, so to say making the inaudible audible.

12.3 Methods of Audification

As with any sonification technique, the overall goal of data audification is to enable us to listen to the results of scientific measurements or simulations in order to make sense of them. As described in the introduction of this chapter, the particularity of audification lies in the fact that the data analysis is delegated almost completely to the human auditory sense. In the case of audification, we therefore try to minimize the transformations of data prior to listening in order to be able to perceive them as far as possible in their “raw” state (i.e., in the form they have been acquired). This chapter refers to these transformations as *signal conditioning* to underline the difference with parameter mapping sonification.

12.3.1 Analytic Listening

Audification is especially useful in cases where numerical data analysis methods fail or are significantly outperformed by the analytical capabilities of the human auditory system. In these cases, it cannot be decided which aspects of the data contain the information we may be interested in. By engaging with the data in a process of analytic listening, patterns may emerge which are otherwise undetectable. Listening to data audifications is a demanding task that needs training and experience. Any approach to audification has to support this task in the best possible way. This implies that all stages of the audification process have to be made explicit such that the listener can determine their influence on the perception of the data at any time. It is crucial to understand which aspects of a sound may stem from which stage of the audification process. This is the only way to distinguish features in the data set from artifacts inevitably introduced by any process of observation.

12.3.2 The Audification Process Model

The various technical aspects of audification are presented in Figure 12.3 according to a model of a typical audification process, which may be divided into three sequential stages: 1) data acquisition, 2) signal conditioning, and 3) sound projection. Each of these stages will be discussed in detail below once the overall constraints informing this process have been clarified. The goal of the audification process is to format the data in such a way that it is best exposed to the analytical capabilities of the human auditory system. The overall design of the audification process and the choices to be taken at each of its stages depend on a) the characteristics of the auditory system, b) the characteristics of the data set, and c) the questions that drive our analysis (cf. Fig. 12.3).

Characteristics of the Human Auditory System

The characteristics of the human auditory system most relevant to audification are its frequency range, frequency resolution, temporal resolution, dynamic range, masking effects, and

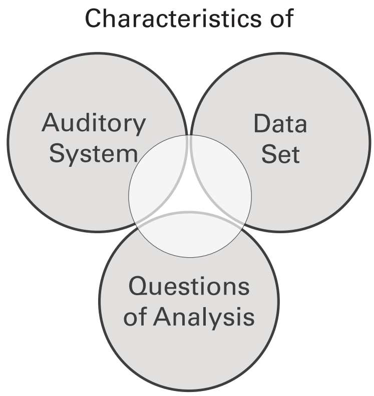

Figure 12.3: Aspects Informing the Audification Process

the different types of detection and discrimination tasks the auditory system has specialized in during its evolution (see chapters 3 and 4 for details).

The ear is sensitive over a *frequency range* of about 20 Hz to 20 kHz. Especially the upper limit of this range varies significantly with age and gender. Young individuals may be able to detect signals far above 20 kHz. Due to age-related hearing loss (presbycusis) the upper limit drops as much as an octave with progressing age, but there are significant differences between individuals. Noise-induced hearing impairment may further reduce the upper limit. It seems relatively safe to assume that even older individuals with typical presbycusis and no additional impairment hear up to 10 kHz. In practical applications the lower end of the frequency range is limited by the typical reproduction system by yet another octave. Thus, from the theoretical bandwidth of 10 octaves, only 8 can be used in practical applications (40 Hz – 10 kHz). The conditioning of a data set has to respect these bandwidth limitations, i.e., interesting signal components have to be transformed into this range. The *frequency resolution* of the ear is about 4 Hz in the middle range (1 – 2 kHz), i.e., smaller changes in frequency in a data set cannot be detected. The *temporal resolution* with which the ear can detect individual events in a signal is linked to the lower limit of the frequency range and lies somewhere between 20 and 50 ms. This limitation has to be taken into account if temporal structures in a data set are supposed to be detected through audification.

The *dynamic range* of the human auditory system (see chapter 3) is capable of covering an extent of about 120 dB at middle frequencies (around 1 kHz). This range reflects the level difference between the threshold of hearing (0 dB SPL[10] @ 1 kHz) and the threshold of pain (120 dB SPL @ 1 kHz). Evidently, this range is not available in practical applications. The lower end is limited by background noise (as much as 50 dBA[11] for a typical office space or as little as 20 dBA for a professional studio), and the upper end by the threshold of comfort (around 100 dBA). This limits the usable dynamic range in a practical application to about

[10]SPL stands for Sound Pressure Level. This is an absolute level measured in decibels (dB) refering to the threshold of hearing at 1 kHz, which is defined as the sound pressure of 20 μPa RMS = 0 dB SPL.
[11]The postfix A indicates an A-weighted sound pressure level generally used for noise measurements.

50 to 80 dB. If a data set exhibits a substantially larger dynamic range, the conditioning will have to include dynamic compression.

Another very important characteristic of the human ear is *masking*, occurring in different forms, which may render certain signal components inaudible in the presence of others. These effects are difficult to quantify, and this chapter can only give an overview of the typical situations where masking occurs. If two sounds occur simultaneously and one is masked by the other, this effect is referred to as *simultaneous masking*. A sound close in frequency to a more intense sound is more likely to be masked than if it is far apart in frequency. Masking not only occurs in the frequency domain but also in time. There are two types of *temporal masking*. A weak sound occurring soon after the end of a more intense sound is masked by the more intense sound (forward masking). Even a weak sound appearing just before a more intense sound can be masked by the higher intensity sound (backward masking). The amount of masking occurring in a particular signal depends on the structure of the signal and can only be predicted by a psychoacoustic model (cf. chapter 3). Since the masking effects are frequency- and time-dependent, different forms of conditioning (especially various forms of Time Axis Manipulation) may dramatically change the amount of masking that occurs in a data set.

Besides the characteristics discussed so far, the ear exhibits a number of very specialized capacities to detect and discriminate signals. These capabilities evolved out of any animal's need to determine the sources of sounds in their environment for reasons of survival. Thus, the ear is specialized in grouping elements in complex sounds and attributing these groups to sources. Typically, the grouping is based on discovering structural invariants in the time-frequency patterns of sound. Such invariants are the result of the physical constraints that vibrating objects obey, and which result in clear signatures in the sound. The ear can be thought of as constantly trying to build and verify these grouping hypotheses. This is why it can also deal with situations quite efficiently where the clear signatures of mechanically produced sounds are missing. The ability to scrutinize abstract sound through analytical listening and make sense of it can be considered the basis of data audification. The perceptual strategies listeners employ in this process are generally referred to as Auditory Scene Analysis [9], which is concerned with sequential and simultaneous integration (perceptual fusion) and the various interactions between these two basic types of perceptual grouping.

Characteristics of the Data set

The decisions to be taken at the different stages of the audification process also depend, to a large extent, on the type of data to be explored. As described in the introduction of this chapter, we can distinguish four types of data used in audification. In all cases, it is important to know how the data sets were acquired and which aspects of which processes they are meant to represent. In the first three cases, the data is derived from physical processes, implying that some kind of sensor[12] is used for data acquisition. Usually, these sensors produce an electric current, which has to be amplified and which is–nowadays–directly converted to a digital signal (digitization, A/D conversion) and stored in this form. The quality of a digital signal (i.e., how accurately it represents the physical property measured) mainly depends on the performance of the sensor, the amplifier, and the A/D converter.

[12] A sensor is a transducer which converts a physical condition into an analog electric signal (e.g., microphone, hydrophone, seismometer, thermometer, barometer, EEG electrode, VLF receiver, image sensor).

Every data acquisition process is afflicted with the introduction of artifacts. Typical artifacts are thermal noise[13], mains hum, RF interference, as well as linear and non-linear distortion. If an analog storage medium is used before digitization (e.g., a tape recorder), yet another source of artifacts is introduced. Apart from the technical artifacts introduced by the data acquisition process, other disturbing signal components are usually present in the acquired data (e.g., different types of environmental noise such as ocean waves and traffic noise in seismograms, or as muscle signals and DC offsets and drifts in EEG registrations).

In the audification process the decision needs to be made whether to remove what is considered an artifact before listening to the data, or if this task is delegated to the auditory system, which is often much better in doing so than a pre-applied conditioning algorithm. As discussed above, artifacts are only a problem for the analytical listening process if they are not identified as such, and if they cannot be attributed to a defined stage of the audification process. Therefore, the listener's ears need training to clearly identify the artifacts of each stage (e.g., environmental noise, sensor non-linearities, amplifier noise[14], mains hum, quantization noise, aliasing, data compression artifacts). This is comparatively easy in the case of acquiring data from a physical process. If the data to be audified stems from an "abstract" process, such as a numerical simulation or a collection of numerical data representing social or economic quantities (e.g., stock market data), it is much more difficult to decide what is an artifact. In this case, it is important to understand the underlying process as much as possible (e.g., estimate its bandwidth in order to choose an adequate sampling rate). If it is not possible to obtain the necessary insight, then the listener has to keep in mind that the acquired data may appear substantially obscured.

Questions that Drive the Analysis

Another important aspect informing the audification process lies in the questions to be answered through listening to the data. There are situations where this question cannot be answered precisely and thus audification is used exactly for that reason – because it is not yet known what is being searched for. In this case, users want to quickly play a potentially very large data set and then refine their choice of conditioning depending on what they find. Such an approach may be useful when employing audification to quality control of data. Another task may consist in trying to categorize data quickly. This may require listening to the same data set with different conditioning options before being able to take the decision to which category a data set belongs. A typical question to be answered by audification is whether there is some structure in a very noisy data set. Also, in this case, trying a wide range of conditioning options (e.g., different filters and Time Axis Manipulations) may help to answer this question. If users are looking for temporal segmentation of the data, they may benefit from using conditioning options that enhance the contrast in the data, or they may choose a Time Axis Manipulation that does not affect the pitch. If the goal is to find structures in the frequency domain, the signal can be conditioned such that tonal signal components are amplified and noise is rejected – a kind of spectral contrast enhancement, for example implemented as a multi-band noise gate. From this brief overview of the possible questions

[13]Typically induced directly by Brownian motion in resistors

[14]From recent efforts in quality control of biomedical signals we know that in certain frequency bands the SNR of EEG signals is less than 20 dB due to amplifier noise. Training the ear to distinguish the amplifier noise from the neuronal noise is essential in this situation.

that drive data audification it can be seen that special signal conditioning tools may need to be developed in particular cases. But there exists at least a set of general conditioning tools (described below), which has proven to be useful in most audification tasks.

12.3.3 Stages of the Audification Process

The quality of an audification is determined to a large extent by the quality of the signal to be displayed as sound. This is why the systemic constraints of digital signal representation are discussed in some detail in the section on *data acquisition*. Although this stage may not be under the control of the person performing the audification, it is essential to fully understand what happens at this stage for reasons of transparency. Evidently, the *signal conditioning* stage has another important influence on the signal quality. Most of the audification examples accessible on the Internet today suffer from severe distortions introduced at this stage, showing that there is little or no awareness of this problem. In the audio domain, the problems of data acquisition (sound recording) and conditioning (analog and digital audio signal processing) are much more easily detected because in this case listening to the signals is the main purpose. In other domains such problems are usually only discovered once audification is used. Another oft-neglected aspect of the audification process concerns the interface to our ears – the *sound projection* (just think of the typical speakers used with desktop or laptop computers, or the computer-induced background noise in an average office space). According to the proverb "a chain is only as strong as its weakest link", this stage is as important as the preceding ones.

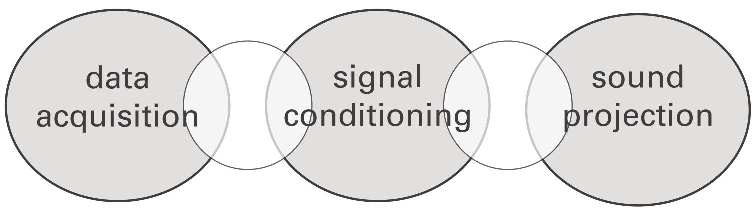

Figure 12.4: Stages of the Audification Process

The Data Acquisition Stage

Not every data set is suitable for audification. An important prerequisite for audification is that the data has the form of – or can be transformed into – a signal. Since input and output data of all audification processes are signals, this section now briefly recalls the foundations of information theory and the sampling process. Most signals of interest for audification are modeled as functions of time or position. A function is a relation, where each element of a set, called *domain* (or input), is associated with a unique element of another set, called *codomain* (or output). The set of all actual output values of a function is called its range. As an example, consider an acoustic signal. Its domain is time and its codomain is air pressure. Its range are all the air pressure values that can be recorded, for instance, with a given microphone.

There are analog and digital signals. Analog signals are continuous, whereas digital signals are discrete and quantized. Analog signals can be converted to digital signals through a process called sampling. Electric signals produced by sensors are discretized and quantized by an analog-to-digital converter (ADC) which implements the sampling process producing the digital signal. Digital signals can also be produced directly by numerical simulations. In both cases the sampling theorem has to be respected. It states: when sampling a signal in time, the sampling frequency must be greater than twice the bandwidth of the input signal in order to be able to reconstruct the original correctly from the sampled version (this reconstruction is essentially what we are doing when audifying data). If this condition is not met, signal components with a frequency above the Nyquist frequency (i.e., half the sampling frequency) will be mirrored at the Nyquist frequency. This artifact is called aliasing or foldover and introduces components into the digital signal which are not present in the analog signal. Similar effects arise in numerical simulations when the time resolution of the simulation is chosen inadequately. Aliasing can be avoided by setting the sampling frequency to cover the full bandwidth of a signal or by limiting the bandwidth with a low-pass filter (anti-aliasing filter) (see chapter 9). High-quality ADCs are equipped with such filters, thus aliasing is usually not a big problem with sampled analog signals. It is much more difficult to avoid (or even to detect) with digitally produced signals, as it is sometimes very hard to estimate the bandwidth of the numerical process. Sparsely sampled data sets, which often also exhibit jitter and missing values, such as historic barometric or temperature data, pose a particular problem to audification. The audible artifacts resulting from the different violations of the sampling theorem need special attention on a case-by-case basis, as a generalization of these effects is impossible.

Another source of errors in digital signals is the quantization of the sample values which adds quantization noise to the signal. The signal-to-noise ratio (SNR) is a measure of the signal quality that can be achieved with a certain level of quantization. It depends on the number of bits used to encode the sample value and can be computed with the following formula (assuming PCM[15] encoding):

$$\text{SNR} = N \cdot 20 \cdot \log_{10}(2), \text{ where } N \text{ is the number of bits.} \tag{1}$$

Consider the following example: according to this formula, a 16-bit digital audio signal has a theoretical SNR of 96.33 dB. But this SNR is only reached for a signal with the highest representable level, i.e., when all bits are used for the quantization. Signal parts with lower levels also have a lower SNR. This is one of the reasons why audio signal encoding moved from 16 to 24 bits in the last decade. This results not only in a better SNR, but also more headroom when recording. Headroom is the margin between the average recording level and the maximum recordable level. A larger headroom prevents signal clipping caused by unexpected signals peaks. Clipping occurs when the signal to be encoded exceeds the quantization range. Like aliasing, clipping also adds signal components to the digital signal which are not present in the analog signal (non-linear distortion). All these aspects of sampling discussed for audio signal here apply equally well to any other type of signal acquisition (e.g., seismic or electromagnetic signals).

[15] Pulse-Code Modulation (PCM) is the standard method used to digitally represent sampled analog signals.

The Signal Conditioning Stage

In the following paragraphs a minimum set of standard signal conditioning tools is introduced in a tutorial style. At the signal conditioning stage it can be assumed that the acquired data is represented as a digital signal with a defined sampling rate *sr*, a bit-depth q. The first example assumes an EEG registration with *sr* = 250 Hz, q = 16 bits.

The Best Conditioning is No Conditioning In the simplest case, no signal conditioning at all is needed to audify a data set. For our EEG signal, this would mean playing it back unchanged at a standard audio sampling rate. As the audio rate is significantly higher than the data rate, this operation would time-compress the signal by a factor equal to the quotient of audio and data rate. For an audio rate of 44.1 kHz, the compression factor would amount to $44100/250 = 176.4$, and so 1 minute of EEG data could be listened to in about 3.4 seconds. Such an audification would typically be useful in a screening or quality control task, for example looking for special events (e.g., electrodes loosening or an epileptic seizure) in a large data set (e.g., in a 24 h registration, which could be scanned in a little over 8 minutes). Evidently, all frequencies in the EEG signals would be transposed up by the same factor (i.e., about 7 1/2 octaves). This means that, for instance the so-called *alpha waves*, which occur in the EEG signal in a frequency range of 8–13 Hz, would become audible in a range of about 1.4–2.3 kHz, which happens to fall into the region where the human ear is most sensitive.

Next Best is Resampling To illustrate a simple case of conditioning by resampling, imagine that the source data are elephant calls, recorded with a microphone and a DAT recorder. As one would expect, typical elephant calls have a very low fundamental frequency (between about 15 and 25 Hz). In order to decide how many individuals are calling on the recording, it must be transposed to a frequency range in which our ears are able to separate the calls. At the original speed, it is only possible to hear an unclear mumbling which all sounds alike. If the sounds had been recorded on an analog tape recorder, the tape could have been played back at double speed and the problem would have been (almost) solved. Eventually, it turns out that a transposition by a factor of 3 moves the elephants' fundamental and the formant frequencies into a range convenient for our ears to easily attribute the calls to different individuals. In the digital domain such a transposition is accomplished by a process called resampling or bandlimited interpolation[16]. This process, which is based on the sampling theorem[17] and the cardinal series, may add samples to a signal by means of interpolation or remove samples from it by reducing its bandwidth and resampling it. In the second case, information is lost since fast fluctuations in the signal (high frequencies) have to be suppressed to avoid aliasing (as discussed above). On the above recording this would concern the birdcalls that were recorded together with the elephants. As they have very high fundamental frequencies, they would have folded over if the bandwidth of the signal were not reduced by low-pass filtering before decimating it. The information loss in the upper part of the spectrum is not critical in this case since the partials of the elephant calls do not reach that far up and we are not interested in the birds.

[16]cf. http://ccrma.stanford.edu/~jos/resample/resample.html (accessed Jan 30 2011)
[17]cf. http://en.wikipedia.org/wiki/Nyquist-Shannon_sampling_theorem (accessed Jan 30 2011)

Filtering is Useful and Mostly Harmless Filtering is a general-purpose conditioning tool useful in various situations. It introduces linear distortion in the signal, i.e., it changes the levels of existing signal components but does not add new components as non-linear distortion does. As an example, imagine that the abovementioned EEG signal was recorded in Europe, where the power lines are operated at a frequency of 50 Hz. With EEG registration it is almost impossible to avoid recording mains hum together with the brain waves. In the earlier example, the artifact would appear as a very annoying sinusoidal parasite at 8820 Hz (50 Hz $\times$ 176.4). Applying a notch filter[18] tuned to this frequency will remove the artifact without interfering too much with other frequency regions. We could solve this problem also by deciding to ignore all frequencies above 40 Hz, which is a relatively sensible assumption for most EEG signals. In this case, we would use a higher-order low-pass filter with a cut-off frequency of 40 Hz. This would attenuate frequencies above 40 Hz (the attenuation having reached -3 dB at the cut-off frequency). Often, filtering is combined with resampling. By resampling the EEG signal by a factor of 3, the original bandwidth of 125 Hz would be reduced to 41.6 Hz. Playing this signal at 44.1 kHz results in a time-compression factor of almost 530 (i.e., a 24 h registration could be auditioned in less than 3 minutes – assuming a young pair of ears that can still hear up to 20 kHz). High-pass filters are used to limit the lower end of the frequency range of a signal. This is typically needed to remove a DC offset or drift from the signal. Another very interesting use of filtering is to control the masking effect. When removing strong signal components with one or more band-pass filters, lower-level components in the surroundings of the attenuated regions will exceed the altered masking threshold and will thus become audible.

Compress Only if there is No Other Way Dynamic compression is needed when signals show large level variations and when the very loud and the very soft parts should both be made audible. As dynamic compression adds non-linear distortion to the signal, it should only be used if there is no other way (e.g., improving the listening conditions). A typical compressor reduces the dynamic range of a signal if it becomes louder than a set *threshold* value. The amount of level reduction is usually determined by a *ratio* control. For instance, with a ratio of 5:1, if the input level is 5 dB above threshold, the signal will be reduced so that the output level will be only 1 dB above threshold. Compressors usually have controls to set the speed with which they respond to changes in input level (known as *attack time*) and how fast they return to neutral operation once the input level falls below threshold (known as *release time*). Dynamic compression may change the character of a signal in subtle to quite drastic ways depending on the settings used. A typical case for dynamic compression in audification involves seismic signals, which exhibit a very large dynamic range and are therefore usually quantized with 24 or even 32 bits (representing 144 resp. 192 dB of theoretical dynamic range). In rare cases dynamic expansion – the inverse process of dynamic compression – may also be useful when a signal exhibits only very small differences in level and a contrast enhancement is needed to better detect changes.

Use Special Tools in Special Cases Only Besides the conditioning techniques described so far, any imaginable signal-processing algorithm may prove useful for a particular audification task. The simplest imaginable conditioning can be seen in time reversal. Due to

[18]A notch filter (also known as band-stop or band-rejection filter) is a special kind of filter that suppresses frequencies only in an extremely small frequency range.

the asymmetry of temporal masking, the listener may detect more (or other) details in the time-reversed version of a signal. Most of the common audio engineering tools such as gates, noise reduction, frequency shifters, etc. may be adequate in special cases. Reverberation, for instance, another common audio engineering tool, may be helpful in situations where very short transient signals are to be audified. Reverberation prolongs the transients in a way that is rather neutral (if a good algorithm with a defined frequency response is used) and which is familiar to our ears. This effect is perceived very dramatically when using headphones for sound projection, since then the influence of room acoustics of the reproduction space is suppressed and different transients may sound quite alike. Once reverberation is added, they can be distinguished much more easily. Non-linear techniques with drastic effects on the signal, such as ring modulation, which is sometimes used for frequency doubling [25], should best be avoided. The spectral side-effects of these techniques are very hard to control and cannot be predicted easily by the human auditory system. But sometimes even such techniques are useful as in heterodyning bat detectors, which use them due to real-time constraints[19].

Making Use of Advanced Spectral Processing Tools Analysis/resynthesis algorithms, such as the phase vocoder [20] and its recent improvements [50, 12], are an interesting class of signal conditioning tools. The phase vocoder allows for time stretching without affecting the pitch, and pitch scaling without affecting the signal duration. In the case of signal resampling discussed above, time and pitch manipulations are always linked (time compression scales pitches up, time stretching scales pitches down). Despite the improvements of the phase vocoder algorithms, they still produce audible artifacts, but they can be easily detected as such. A special case of improvement concerns the treatment of transient portions in a signal, which cannot be time-stretched by definition. The algorithm [53] detects the transients and treats them specially. Independent control of time and pitch is a powerful tool for audification as the signal can be adapted to the characteristics of the auditory system. Signals with rhythmical structures too fast to perceive in detail can be slowed down without making them unperceivable due to low pitch. The spectral structure of a signal can be transposed to regions where the ear is especially efficient in building gestalts without changing the temporal structure of the signal.

The Sound Projection Stage

In the sound projection stage the conditioned signals may be mapped spatially. This is of interest when audifying more than one channel at a time and when the data set exhibits a spatial structure which should be displayed. Another reason for spatial rendering is to exploit the listener's spatial hearing capabilities in order to assist in the auditory gestalt formation, e.g., through spatial unmasking and spatial auditory scene analysis.

The achievable quality of spatial rendering depends on the sound projection setup and the algorithms used to drive a particular setup. Evidently, the best rendering quality is reached if one speaker is used per signal, in which case rendering consists of a simple assignment of signals to loudspeaker channels. It is best to distinguish between rendering for various configurations of loudspeakers as well as for stereo headphones. Rendering for headphones

[19] Cf. `http://www.bats.org.uk/pages/bat_detectors.html` (accessed Jan 30 2011)

may use binaural synthesis employing head-related transfer functions (HRTF) and eventually room acoustic modeling to position sound sources in a virtual sound scene. The localization quality of a binaural display can be enhanced by employing a head tracker to compensate for the user's head movements. Tracked binaural synthesis allows for a stable rendering of sound source locations (i.e., the sound scene does not move with the user's head movement). Under special conditions, binaural rendering may also be used with stereo loudspeaker projection. In this case, the location of the user's head is constrained to a small region. A typical situation for this type of projection is being seated in front of a computer screen. In order to improve localization of sources from behind the user, a second pair of loudspeakers may be used, placed behind the user's head. In this case, an extension of binaural rendering using crosstalk cancellation has to be employed. If head tracking is available, the crosstalk cancellation can be made dynamic, thus enlarging the available sweet spot considerably [36].

Various rendering techniques for multichannel loudspeaker setups are available ranging from simple panning techniques (e.g., Vector Base Amplitude Panning / VBAP [51]) via Higher-Order Ambisonics (HOA [38]) to wave field synthesis (WFS [7]). WFS is a technique which requires a large number of loudspeakers (up to hundreds) but can achieve a very high quality of rendering – but at a very high cost. HOA and simpler amplitude panning techniques are used for loudspeaker setups in two dimensions (rendering of sources in a plane around the listener, e.g., with a ring of 5 or 8 speakers) and three dimensions (rendering of sources with elevation, e.g., a dome of 24 speakers). Standard formats such as stereo or 5.1 surround may also be used for reasons of compatibility and availability.

12.4 Audification now (1992-today)

The first ICAD in 1992, organized by Gregory Kramer, was not only a kick-off for the succeeding conferences, but also the formation of sonification research as a new discipline. It is therefore appropriate to assume a *caesura* here and to give the audification research after the first ICAD another section in this chapter.

In 1992, audification as a sonification technique received its name [32, p. xxvii] and its definition was quickly refined.[20] As a result professional investigation improved and empirically reliable audification research could develop, especially in the context of the yearly ICAD. Nevertheless, papers explicitly on audification are – as we will see – rare. Even today, audification is usually used only as a mock-up for sonification research, and is not described (or only without much detail) in publications of the ICAD proceedings. Much more vivid are the amateurs' applications of audification to all kinds of data, mostly for science popularization or amusement. This boom of auditory bricolage certainly relates to the development of the computer as an audiovisual display. Notably, the function of audification as a gimmick in mathematical visualization software, such as *Mathematica* ("Play[...]") or *MathTrax* should not be underestimated.

The following paragraphs summarize the few serious scientific works on audification and try to give a little overview of further ideas and applications in different areas of less scientific claim.

[20] Cf. also the *Sonification Report* of 1997, `http://www.icad.org/websiteV2.0/References/nsf.html`, accessed Jan 30 2011 and [27, pp. 35-40]

12.4.1 Scientific Examples

The most recent profound investigation of audification as a technique was carried out by Sandra Pauletto and Andy Hunt (2005), in which they tried to understand the advantages and disadvantages of audification in general [48]. There are many other papers which compare auditory and visual representations and also deal briefly with audification, although they usually do not go into detail. As representative of this type of work, the reader's attention is drawn to [37] and [6].

(i) Medicine: From the abovementioned examples of early audifications (cf. section 12.2), the stethoscope is still in use with great success, even though it functions no longer as a scientific proof, but more as a demonstration of argument. The idea of audifying EEG and nerve measurements developed in the 19$^{\text{th}}$ century was almost forgotten for more than a century but is now making a comeback: people such as Gerold Baier and Thomas Hermann are seriously investigating EEG sonifications, even though the technique of audification plays a minor role compared to parameter mapping (sound example **S12.15**).[21] One special publication in this field worth mentioning was delivered by Jesus Olivan, Bob Kemp and Marco Roessen in 2004 [47], in which they used audification to investigate EEG sleep recordings. Besides EEG data, heart rate variability has also attracted the interest of sonification researchers using audification [5].

(ii) Seismology: One area where audification finds a highly promising application is seismology. Chris Hayward brought up the topic at the first ICAD in 1992 (sound example **S12.16**) [25]. Co-operating closely with Gregory Kramer, he carried out an extensive and diligent investigation of the topic and was the first to examine the overall potential of audification in the area of seismics and seismology. Today, there are people still researching in the area, such as Frank Scherbaum from the University of Potsdam or Florian Dombois (sound example **S12.17**), who established what is known today as "Auditory Seismology".[22] And in 2008, Meier and Saranti presented some sonic explorations with seismic data [42].

(iii) Physics: Apart from Hayward's, there are two more examples of the use of audification in the proceedings of ICAD 1992: one on computational fluid dynamics data by McCabe and Rangwalla [41], and one on chaotic attractors by Mayer-Kress, Bargar and Insook [40]. The propositions of these authors have not been followed too far, as far as is known. An exception was the research group of Hans Diebner at ZKM Karlsruhe (D), who audified chaos of all kinds, displaying it at some of the art shows of ZKM. A more famous example of successful audification in Physics is the discovery of quantum oscillations in He-3 atoms, where Pereverzev et al., in 1997, found the searched-for frequency by listening directly to measured data [49]. Another interesting application in the nano sector was presented in 2004 at the Interactive Sonification workshop in Bielefeld, where Martini et al. used audification for "fishing" single atoms [39].

(iv) Stock market: In the area of *abstract data,* the stock market has a major attraction for obvious reasons. Unfortunately, these research results are usually not published. S. P. Frysinger's work of 1990 used audification of stock market data [23], which has been

[21] Cf. [4] and [3]; cf. also [27]; a lot of listening examples can be found at http://www.sonifyer.org/sound/eeg/ (accessed Jan 30 2011)

[22] For an overview of the history of Auditory Seismology see [15]. For further research look at [13], [14] or [16]. An updated publication list and several sound examples can be found at http://www.auditory-seismology.org (accessed Jan 30 2011).

evaluated by Keith Nesbitt and Stephen Barrass [45, 46]. And David Worrall published a paper with a fine overview on the area of sonifying stock market data [62].

(v) Statistics: There was also some interesting work done in the area of high order statistics by Frauenberger, de Campo and Eckel, analyzing statistical properties of time-series data by auditory means [22]. Audification was mainly used to judge skewness and kurtosis.

All in all, one can say that audification has become, over the last ten years, quite common in the scientific community, due to computer programs such as *Mathematica* or *MathTrax*, that easily audify all kinds of data. Nevertheless, the number of researchers that assume audification as a reasonable research method is still small. It is seen more as a nice gimmick in the popularization of science, and here one can find innumerable websites in all scientific domains:

For example, famous are the transposed whale chantings that have been recorded and sold as CDs all over the world. A project, "The dark side of the cell", also received a lot of publicity because membrane oscillations of living cells had been audified.[23] In astronomy, NASA has used audification over the last several years to portray their results in a novel way in order to attract the interest of the general public. In 2004, when Cassini flew in and out of Saturn's ring plane, the press release also contained an audification of radio and plasma wave measurements.[24] Similar audifications can also be found of Titan or general solar wind registrations.[25] NASA even installed an online web radio of real-time audifications of VLF recordings at the Marshall Space Flight Center as part of their teaching program, INSPIRE.[26] Also, astrophysicists, such as Mark Whittle, used audio to successfully market their big bang calculations not only in the scientific community.[27] And there are sound-lovers, like Don Gurnett, who are publishing all kinds of sonifications of astrophysical data on the web.[28] According to Tim O'Brien, astronomers are increasingly listening into stars and other space sounds because "[i]t's interesting in itself [and i]t's also scientifically useful."[29] In Geophysics, we also find educational uses of audification. The USGS and John Louie set up two reasonable websites with audifications of earthquake registrations.[30]

12.4.2 Artistic Examples

The interest in creating new sounds, especially in computer music, led many musicians into audification of all kinds of data. But this section highlights only a very few examples of

[23] http://www.darksideofcell.info (accessed Jan 30 2011)

[24] http://www1.nasa.gov/mission_pages/cassini/multimedia/pia06410.html (accessed Jan 30 2011)

[25] http://www.nasa.gov/vision/universe/solarsystem/voyager-sound.html or http://www.esa.int/SPECIALS/Cassini-Huygens/SEM85Q71Y3E_0.html (accessed Jan 30 2011)

[26] http://www.spaceweather.com/glossary/inspire.html resp. http://science.nasa.gov/headlines/y2001/ast19jan_1.htm. Further VLF-recordings can be found at http://www-pw.physics.uiowa.edu/plasma-wave/istp/polar/magnetosound.html (accessed Jan 30 2011)

[27] http://www.astro.virginia.edu/~dmw8f/index.php, see also http://staff.washington.edu/seymour/altvw104.html (accessed Jan 30 2011)

[28] http://www-pw.physics.uiowa.edu/space-audio/ (accessed Jan 30 2011)

[29] http://news.bbc.co.uk/2/hi/science/nature/7687286.stm (accessed Jan 30 2011)

[30] John Louie http://crack.seismo.unr.edu/ftp/pub/louie/sounds/ and USGS http://quake.wr.usgs.gov/info/listen/. See also the amateur site of Mauro Mariotti http://mariottim.interfree.it/doc12_e.htm (accessed Jan 30 2011)

intentional use of audification in the last years, where the resulting sounds have been used without major aesthetic manipulations. Left out is the huge body of works which transform mechanical waves directly into sound waves without any transposition.

An interesting project is "According to Scripture", by Paul DeMarini, who revitalized, in 2002, a series of visual waveform diagrams from the 19th century, that were not registered on a phonograph but drawn directly on paper. Among others, one finds here E.W. Scripture's famous notations from 1853–1890 digitized and reaudified, giving the ear access to the oldest sound registrations ever made.

Christina Kubisch's famous project, "Electrical Walks" (sound example **S12.18**), was first shown in 2004. The visitor receives a headphone that audifies electromagnetic induction from the surroundings, and a city map with a suggested tour.[31] An enormous sound space is opened up by the prepared headphone that restructures the topology of the city.[32]

In 2004, the Australian group *radioqualia* displayed their piece "Radio Astronomy" at the Ars Electronica Festival in Linz (A), where real-time VLF recordings could be listened to. It was a network project, working together with the Windward Community College Radio Observatory in Hawaii, USA, NASA's Radio Jove network, the Ventspils International Radio Astronomy Centre in Latvia and the cultural center RIXC from Riga, Latvia.

Under the motto "art as research", several sound installations of seismological data by Florian Dombois have been shown at Cologne Gallery Haferkamp in 2003 and 2006[33] and [34], and also at Gallery gelbe MUSIK in Berlin 2009 etc. Here, one could listen, for example in "Circum Pacific" (sound example **S12.19**), to five seismic stations monitoring the seismic activity around the pacific plate.

Also at Ars Electronica, the following year 2005, "G-Player" (sound example **S12.20**) was shown, an artwork by Jens Brand from Germany. He uses a topographic model of the earth and imagines satellites behaving like the needle of a record player, so that a topographic cross section (following the flight route) can be directly audified and listened to [30, p. 342f.].

12.5 Conclusion: What audification should be used for

Before applying any sonification technique to a data set, the focus of interest should be considered. Sonification is not a "universal remedy", and if, for example, a visualization has been successful, why invest in another data transformation? One should be aware that listening is quite different from looking, and one should therefore approach a data set acoustically mainly in those cases where visualization usually fails.

Within the sonification techniques, audification is surely the most direct and simple to handle. No sound engines are needed, no instruments, no libraries of samples, no acoustic inputs. Audification, therefore, always bears a fundamental surprise, and the characteristics of the acoustic results are usually hard to foresee. But not every data set is suitable; it should fulfill some preconditions:

1. **number of samples:** Audification requires large quantities of data. The usual resolu-

[31] Cf. the interview in [11].
[32] http://www.cabinetmagazine.org/issues/21/kubisch.php (accessed Jan 30 2011)
[33] Cf. http://www.rachelhaferkamp.eu (accessed Jan 30 2011)

tion of an audio track is 44100 samples per second. To apply audification to a data set with listenable output, it should contain at least a few thousands of samples.

2. **wave-like signals:** Audification always needs at least one-dimensional data sets interpreted as a time series signal. The data should not have too many breaks or dropouts, and a round curve-like shape usually delivers the best output. The audification approach is most promising when dealing with data following physical laws, especially from elastomechanics. In these cases, audification can be assumed as an extension of the ear, conquering frequencies outside the usual range of perception, comparable to thermo- or x-ray-photography.
3. **complexity:** The ear can cope easily with complex sounds, and audification affords this opportunity much more easily than any other sonification technique. One should, therefore, always supply audification first with all the complexity that is in the data before doing any reduction or filtering.
4. **signal to noise ratio:** The cocktail-party-effect – i.e., the ability to focus one's listening attention on a single talker among a mixture of conversations and background noises, ignoring other conversations – is quite well known. Due to this effect, audification is a good approach for finding hidden signals in a noisy data record, assuming the audified data to be a mixture of unknown sources. The ear is trained to separate different sources, and therefore audification can lead to the discovery of unexpected signals or structural invariants in the time-frequency patterns of sound or all kinds of implausible artifacts that are interspersing.
5. **subtle changes:** Slow changes in data characteristics (e.g., frequencies or rhythms) can be obtained mostly easily due to the high flow rate of samples, whereas driftings are often difficult to recognize and often get lost by high-pass filtering.
6. **simultaneousness of rhythmical patterns:** Cross-correlations usually need a lot of calculation time and are visually difficult to obtain. Here, audification gives good access to follow several signals simultaneously and can demonstrate whether the rhythm is in or out of synchrony.
7. **data screening:** A suitable area for audification are all kinds of screening tasks. It is a good approach for getting a quick overview of the characteristics of different data sets.
8. **classification:** The ear structures signals very differently from the eye. One can, therefore, use audification also for all questions of classification.

12.6 Towards Better Audification Tools

Until recently, software tools that support the process of analytical listening to the degree required for a successful application of data audification have been lacking. This section presents a few general guidelines for the development of such tools. The recent past has seen several initiatives to create an integrated software environment for sonification.[34] As the field

[34]SonEnvir at IEM in Graz (http://sonenvir.at/), Denkgeräusche at HKB in Bern (http://www.sonifyer.org see also [17]), NASA's xSonify (http://spdf.gsfc.nasa.gov/research/sonification/sonification_software.html), Georgia Tech's Sonification Sandbox http://sonify.psych.gatech.edu/research/sonification_sandbox/. (All accessed Jan 30 2011)

of sonification is vast, and audification still tends to be undervalued, the particular needs of listening to data as directly as possible are only taken into account to a small degree in these development projects.

As the focus is on listening, all graphical data representations supporting this task have to be part of a general-purpose audification tool, under the condition that they can be switched off, i.e., made invisible, in order not to bias listening. Our tool has to allow us to grow with it, to improve our listening skills by using it and to enable us to extend it if we reach the limits of the possibilities foreseen by the developers. One of the most important features of the tool in question is the possibility of quickly checking ad hoc hypotheses about the problem under examination, as users develop in the process of analytical listening. Transparency of operation and the highest possible implementation quality of the employed signal processing algorithms (e.g., resampling) are the other essential features.

Programmability should be integrated on the level of an extension language, either in textual form, as for instance in SuperCollider,[35] or in graphical form like in Max[36] (or best in both). Extensibility is important to develop custom signal conditioning algorithms and for adding data import routines, as they are needed, when audifying data sets in formats not yet supported by the audification tool. As a central element, the tool has to include a multichannel time domain and frequency domain signal editor (with features as in AudioSculpt[37] and STx[38]). It goes without saying that another important feature of our ideal tool is interactivity. Changing conditioning parameters in real time and efficiently browsing large data sets are important prerequisites to work efficiently with audification.

Once a more powerful software environment is settled the real work can start: audifying all kinds of data. We think, that mechanical waves are most promising and especially natural frequencies. Listening to ultrasonic resonances of, for example, fruits, vegetables, cheese[39] up to infrasonic as in sculptures, buildings, bridges, planetary bodies etc.[40]. There is a whole cosmos of neglected sounds, that wait to be investigated.

Bibliography

[1] Adrian, E. D., & Matthews, B. H. C. (1934). The Berger Rhythm: Potential Changes from the Occipital Lobes in Man. *Brain* 57:355–385.

[2] Auenbrugger, L. (1936). On the Percussion of the Chest (transl. by John Forbes). *Bulletin of the History of Medicine* 4:373–404.

[3] Baier, G. (2001). *Rhythmus. Tanz in Körper und Gehirn*. Hamburg: Rowohlt.

[4] Baier, G., Hermann, Th., Lara, O. M. & Müller, M. (2005). Using Sonification to Detect Weak Cross-Correlations in Coupled Excitable Systems. Paper read at ICAD 2005, July 6–9, at Limerick, Ireland.

[5] Ballora, M., Pennycook, B. W., & Glass, L. (2000). Audification of Heart Rhythms in Csound. In *The Csound Book: Perspectives in Software Synthesis, Sound Design, Signal Processing, and Programming*, edited by R. Belanger. Cambridge (MA): MIT Press.

[35] http://supercollider.sourceforge.net/ (accessed Jan 30 2011)

[36] http://cycling74.com/products/maxmspjitter/ (accessed Jan 30 2011)

[37] cf. http://forumnet.ircam.fr/349.html?\&L=1 (accessed Jan 30 2011)

[38] cf. http://www.kfs.oeaw.ac.at/content/blogsection/11/443/lang,8859-1/ (accessed Jan 30 2011)

[39] cf. http://www.sonifyer.org/sound/kartoffel/ or http://www.sonifyer.org/sound/ausserdem/?id=27

[40] cf. e.g. http://www.klangkunstpreis.de/preise_2010.php or several examples in [34]

[6] Barrass, S., & Kramer, G. (1999). Using Sonification. *Multimedia Systems* 7:23–31.

[7] Berkhout, A. J., De Vries, D. & Vogel, P. (1993). Acoustic Control by Wave Field Synthesis. In *Journal of the Acoustical Society of America*, 93:2764–2778

[8] Bernstein, J., & C. Schönlein. (1881). Telephonische Wahrnehmung der Schwankungen des Muskelstromes bei der Contraction. *Sitzungsberichte der Naturforschenden Gesellschaft zu Halle*:18–27.

[9] Bregman, A. S. (1990). *Auditory Scene Analysis: The Perceptual Organization of Sound.* Cambridge, Massachusetts: The MIT Press.

[10] Cocteau, J. (1983). *Kino und Poesie. Notizen*. Frankfurt a. M.: Ullstein.

[11] Cox, Ch. (2006). Invisible Cities: An Interview with Christina Kubisch. In *Cabinet Magazine* 21

[12] Dolson, M. & Laroche, J. (1999). Improved phase vocoder time-scale modification of audio. *IEEE Transactions on Speech and Audio Processing* 7.3:323–332.

[13] Dombois, F. (1999). *Earthquake Sounds. Volume 1: Kobe 16.1.1995, 20:46 ut*. St. Augustin. (Audio CD)

[14] Dombois, F. (2001). Using Audification in Planetary Seismology. Paper read at 2001 International Conference on Auditory Display, July 29–August 1, at Espoo.

[15] Dombois, F. (2002). Auditory Seismology: On Free Oscillations, Focal Mechanisms, Explosions and Synthetic Seismograms. Paper read at 2002 International Conference on Auditory Display, July 2–5, at Kyoto.

[16] Dombois, F. (2006). Reflektierte Phantasie. Vom Erfinden und Erkennen, insbesondere in der Seismologie. *Paragrana* Beiheft 1:109–119.

[17] Dombois, F. (2008a). Sonifikation: Ein Plädoyer, dem naturwissenschaftlichen Verfahren eine kulturhistorische Einschätzung zukommen zu lassen. In *Acoustic Turn* edited by Petra Maria Meyer. Paderborn: Fink, 91–100.

[18] Dombois, F. (2008b). The 'Muscle Telephone': The Undiscovered Start of Audification in the 1870s. In *Sounds of Science – Schall im Labor (1800–1930)* edited by Julia Kursell. Berlin: Max Planck Institute for the History of Science, Preprint 346:41–45.

[19] Dombois, F., Brodwolf, O., Friedli, O., Rennert, I. & König, Th. (2008). SONIFYER – A concept, a Software, a Platform. Paper read at 2008 International Conference on Auditory Display, June 24–27, at Paris.

[20] Flanagan, J. L. & Golden, R. M. (1966). Phase vocoder, *Bell Syst. Tech. J.* 45:1493–1509.

[21] Frantti, G. E., & Leverault, L. A. (1965). Auditory Discrimination of Seismic Signals from Earthquakes and Explosions. *Bulletin of the Seismological Society of America* 55:1–25.

[22] Frauenberger, Ch., de Campo, A. & Eckel, G. (2007). Analysing Time Series Data. Paper read at 2007 International Conference on Auditory Display, June 26–29, at Montreal, Canada.

[23] Frysinger, S. P. (1990). Applied Research in Auditory Data Representation. Paper read at SPIE/SPSE Symposium on Electronic Imaging, February, at Springfield (VA).

[24] Gelatt, R. (1977). *The Fabulous Phonograph 1877–1977*. 2 ed. London: Cassell.

[25] Hayward, Ch. (1994). Listening to the Earth Sing. In *Auditory Display. Sonification, Audification, and Auditory Interfaces*, edited by G. Kramer. Reading: Addison-Wesley, 369–404.

[26] Hermann, L. (1878). Ueber electrophysiologische Verwendung des Telephons. *Archiv für die gesamte Physiologie des Menschen und der Tiere* 16:504–509.

[27] Hermann, Th. (2002). *Sonification for exploratory data analysis*, PhD thesis, Bielefeld University, Bielefeld, available via http://sonification.de/publications/Hermann2002-SFE

[28] Hermann, Th., & Hunt, A. (2004). The Discipline of Interactive Sonification. Proc. Int. Workshop on Interactive Sonification (ISon 2004), Bielefeld, http://interactive-sonification.org/ISon2004/proceedings/papers/HermannHunt2004-TDO.pdf

[29] Hermann, Th., & Hunt, A. (2005). An Introduction to Interactive Sonification. *IEEE Multimedia* 12 (2):20–24.

[30] Kiefer, P. (ed.). (2010). *Klangräume der Kunst.* Heidelberg: Kehrer.

[31] Kittler, F. (1986). *Grammophon. Film. Typewriter*. Berlin: Brinkmann & Bose.

[32] Kramer, G. (1994). An Introduction to Auditory Display. In *Auditory Display: Sonification, Audification and*

Auditory Interfaces, edited by G. Kramer. Reading (MA): Addison-Wesley.

[33] Kramer, G. (ed.). (1994). *Auditory Display: Sonification, Audification and Auditory Interfaces*. Reading (MA): Addison-Wesley.

[34] Kunsthalle Bern (ed.). (2010). *Florian Dombois: What Are the Places of Danger. Works 1999-2009.* Berlin: argobooks.

[35] Laënnec, R. Th. H. (1830). *A Treatise on the Diseases of the Chest and on Mediate Auscultation*. Translated by J. Forbes. 3 ed. New York: Wood, Collins and Hannay.

[36] Lentz, T., Assenmacher, I., & Sokoll, J. (2005). Performance of Spatial Audio Using Dynamic Cross-Talk Cancellation. Proceedings of the 119th Audio Engineering Society Convention, NY.

[37] Madhyastha, T. M., & Reed, D. A. (1995). Data Sonification: Do You See What I Hear? *IEEE Software* 12 (2):45–56.

[38] Malham, D. G. (1999). Higher order Ambisonic systems for the spatialisation of sound. In *Proceedings of the International Computer Music conference, Beijing* 484–487.

[39] Martini, J., Hermann, Th., Anselmetti, D., & Ritter, H. (2004). Interactive Sonification for exploring Single Molecule Properties with AFM based Force Spectroscopy. Paper read at International Workshop on Interactive Sonification, January, 8, at Bielefeld.

[40] Mayer-Kress, G., Bargar, R., & Insook, Ch. (1994). Musical structures in Data from chaotic attractors. In *Auditory Display: Sonification, Audification and Auditory Interfaces*, edited by G. Kramer. Reading, Massachusetts, USA: Addison-Wesley.

[41] McCabe, K., & Rangwalla, A. (1994). Auditory Display of Computational Fluid Dynamics Data. In *Auditory Display: Sonification, Audification and Auditory Interfaces*, edited by G. Kramer. Reading, Massachusetts, USA: Addison-Wesley.

[42] Meier, M., & Saranti, A. (2008). Sonic Explorations with Earthquake Data. Paper read at ICAD 2008, 24.–27.6.2008, at Paris, France.

[43] Moholy-Nagy, L. (1923). Neue Gestaltung in der Musik. *Der Sturm* Juli:102–106.

[44] Moritz, W. (2004). *Optical Poetry. The Life and Work of Oskar Fischinger*. Bloomington: Indiana University Press.

[45] Nesbitt, K., & Barrass, S. (2002). Evaluation of a Multimodal Sonification and Visualisation of Depth of Market Stock Data. Paper read at 2002 International Conference on Auditory Display (ICAD), at Kyoto.

[46] Nesbitt, K., & Barrass, S. (2004). Finding Trading Patterns in Stock Market Data. *IEEE Computer Graphics* 24 (5):45–55.

[47] Olivan, J., Kemp, B. & Roessen, M. (2004). Easy listening to sleep recordings: tools and examples. *Sleep Medicine* 5:601–603.

[48] Pauletto, S., & Hunt, A. (2005). A Comparison of Audio & Visual Analysis of Complex Time-Series Data Sets. Paper read at ICAD 2005, 6.–9.7.2005, at Limerick, Ireland.

[49] Pereverzev, S. V., Loshak, A., Backhaus, S, Davis, J. C., & Packard, R. (1997). Quantum oscillations between two weakly coupled reservoirs of superfluid He-3. *Nature* 388:449–451.

[50] Puckette, M.(1995). Phase-locked vocoder. *Proc. IEEE ASSP Conference on Applications of Signal Processing to Audio and Acoustics* (Mohonk, N.Y.).

[51] Pulkki, V. (1997). Virtual sound source positioning using vector base amplitude panning. In *Journal of the Audio Engineering Society*, 45(6):456–466.

[52] Rilke, R. M. (1987). Ur-Geräusch (1919). In *Sämtliche Werke Bd. 6: Die Aufzeichungen des Malte Laurids Bridge. Prosa 1906–1926*. Frankfurt a. M.: Insel.

[53] Röbel, A. (2003). A new approach to transient processing in the phase vocoder, Proc. of the 6th Int. Conference on Digital Audio Effects (DAFx-03), London, UK.

[54] Russcol, H. (1972). *The Liberation of Sound. An introduction to electronic music*. Englewood Cliffs (NJ): Prentice Hall.

[55] Speeth, S. D. (1961). Seismometer Sounds. *Journal of the Acoustical Society of America* 33:909–916.

[56] Schoon, A. (2011). Datenkörper aushorchen. *kunsttexte.de / Auditive Perspektiven* 2.

[57] Schoon, A., & Dombois, F. (2009). Sonification in Music. Paper read at ICAD 2009, 18.–22.5.2009, Copenhagen, Denmark.

[58] Sterne, J. (2003). *The Audible Past: Cultural origins of sound reproduction*. Durham: Duke University Press.

[59] Volmar, A. (2010). Listening to the Body Electric. Electrophysiology and the Telephone in the Late 19th century. *The Virtual Laboratory,* edited by Max-Planck-Institute for the History of Science (ISSN 1866-4784), http://vlp.mpiwg-berlin.mpg.de/references?id=art76 (accessed Aug 9 2011).

[60] Walker, B., & Kramer, G. (2004). Ecological Psychoacoustics and Auditory Display. In *Ecological psychoacoustics*, edited by J. G. Neuhoff. San Diego: Elsevier.

[61] Wedenskii, N. (1883). Die telephonischen Wirkungen des erregten Nerven. *Centralblatt für die medicinischen Wissenschaften* 31 (26):465-468.

[62] Worrall, D. (2008). The Use of Sonic Articulation in Identifying Correlation in Capital Market Trading Data. Paper read at ICAD 2009, 18.–22.6.2009, at Copenhagen, Denmark.