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Readings

Moving on from Weiser's Vision of Calm Computing:
Engaging UbiComp Experiences

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Abstract. A motivation behind much UbiComp research has been to make our lives convenient, comfortable and informed, following in the footsteps of Weiser's calm computing vision. Three themes that have dominated are context awareness, ambient intelligence and monitoring/tracking. While these avenues of research have been fruitful their accomplishments do not match up to anything like Weiser's world. This paper discusses why this is so and argues that is time for a change of direction in the field. An alternative agenda is outlined that focuses on engaging rather than calming people. Humans are very resourceful at exploiting their environments and extending their capabilities using existing strategies and tools. I describe how pervasive technologies can be added to the mix, outlining three areas of practice where there is much potential for professionals and laypeople alike to combine, adapt and use them in creative and constructive ways.

Keywords: calm computing, Weiser, user experiences, engaged living, UbiComp history, pervasive technologies, proactive computing.

1 Introduction

Mark Weiser's vision of ubiquitous computing has had an enormous impact on the directions that the nascent field of UbiComp has taken. A central thesis was that while "computers for personal use have focused on the excitement of interaction...the most potentially interesting, challenging and profound change implied by the ubiquitous computing era is a focus on calm." [46]. Given the likelihood that computers will be everywhere, in our environments and even embedded in our bodies, he argued that they better "stay out of the way" and not overburden us in our everyday lives. In contrast, his picture of calm technology portrayed a world of serenity, comfort and awareness, where we are kept perpetually informed of what is happening around us, what is going to happen and what has just happened. Information would appear in the centre of our attention when needed and effortlessly disappear into the periphery of our attention when not.

Now regarded as the forefather of UbiComp, Weiser has inspired governments, researchers and developers across the globe. Most prominent was the European Community's Disappearing Computer initiative in the late 90s and early 2000s, that

funded a large number of research projects to investigate how information technology could be diffused into everyday objects and settings and to see how this could lead to

new ways of supporting and enhancing people's lives that went above and beyond what was possible using desktop machines. Other ambitious and far-reaching projects included MIT's Oxygen, HP's CoolTown, IBM's BlueEyes, Philips Vision of the Future and attempts by various telecom companies and academia to create the ultimate 'smart home', e.g., Orange-at-Home and Aware Home. A central aspiration running through these early efforts was that the environment, the home, and our possessions would be aware, adapt and respond to our varying comfort needs, individual moods and information requirements. We would only have to walk into a room, make a gesture or speak aloud and the environment would bend to our will and respond or react as deemed appropriate for that point in time.

Considerable effort has gone into realizing Weiser's vision in terms of developing frameworks, technologies and infrastructures. Proactive computing was put forward as an approach to determine how to program computers to take the initiative to act on people's behalf [43]. The environment has been augmented with various computational resources to provide information and services, when and where desired, with the implicit goal of "assisting everyday life and not overwhelming it" [1]. An assortment of sensors have been experimented with in our homes, hospitals, public buildings, physical environments and even our bodies to detect trends and anomalies, providing a dizzying array of data about our health, movements, changes in the environment and so on. Algorithms have been developed to analyze the data in order for inferences to be made about what actions to take for people. In addition, sensed data is increasingly being used to automate mundane operations and actions that we would have done in our everyday worlds using conventional knobs, buttons and other physical controls. For example, our favorite kind of music or TV show that we like to exercise to will automatically play as we enter a gym. Sensed data is also being used to remind us of things we often forget to do at salient times, such as detecting the absence of milk in the fridge and messaging us to buy a carton when passing the grocery store.

But, as advanced and impressive as these endeavors have been they still do not match up to anything like a world of calm computing. There is an enormous gap between the dream of comfortable, informed and effortless living and the accomplishments of UbiComp research. As pointed out by Greenfield [20] "we simply don't do 'smart' very well yet" because it involves solving very hard artificial intelligence problems that in many ways are more challenging than creating an artificial human [26]. A fundamental stumbling block has been harnessing the huge variability in what people do, their motives for doing it, when they do it and how they do it. Ethnographic studies of how people manage their lives – ranging from those suffering from

Alzheimer's Disease to high-powered professionals – have revealed that the specifics of the context surrounding people's day-to-day living are much more subtle, fluid and idiosyncratic than theories of context have led us to believe [40]. This makes it difficult, if not impossible, to try to implement context in any practical sense and from which to make sensible predictions about what someone is feeling, wanting or needing at a given moment. Hence, while it has been possible to develop a range of simple UbiComp systems that can offer relevant information at opportune moments (e.g., reminding and recommending to us things that are considered useful and important) it is proving to be much more difficult to build truly smart systems that can understand or accurately model people's behaviors, moods and intentions.

The very idea of calm computing has also raised a number of ethical and social concerns. Even if it was possible for Weiser's dream to be fulfilled would we want to live in such a world? In particular, is it desirable to depend on computers to take on our day-to-day decision-making and planning activities? Will our abilities to learn, remember and think for ourselves suffer if we begin to rely increasingly on the environment to do them for us? Furthermore, how do designers decide which activities should be left for humans to control and which are acceptable and valuable for the environment to take over responsibility for?

In this paper I argue that progress in UbiComp research has been hampered by intractable computational and ethical problems and that we need to begin taking stock of both the dream and developments in the field. In particular, we need to rethink the value and role of calm and proactive computing as main driving forces. It is without question that Weiser's enormous legacy will (and should) continue to have an impact on UbiComp developments. However, sufficient time has passed since his untimely death and it should be possible now for researchers to take a critical stance. As part of this exercise, I propose that the field needs to broaden its scope, setting and addressing other goals that are more attainable and down-to-earth. New agendas need also to be outlined that can guide, stimulate and challenge UbiComp (and other) researchers and developers, building upon the growing body of research in the field.

To this end, I propose one such alternative agenda which focuses on designing UbiComp technologies for engaging user experiences. It argues for a significant shift from proactive computing to proactive people; where UbiComp technologies are designed not to do things for people but to engage them more actively in what they currently do. Rather than calm living it promotes engaged living, where technology is designed to enable people to do what they want, need or never even considered before by acting in and upon the environment. Instead of embedding pervasive computing everywhere in the environment it considers how UbiComp technologies can be created as ensembles or ecologies of resources, that can be mobile and/or fixed, to serve specific purposes and be situated in particular places. Furthermore, it argues that peo-

ple rather than computers should take the initiative to be constructive, creative and, ultimately, in control of their interactions with the world – in novel and extensive ways.

While this agenda might appear to be a regressive step and even an anathema to some ardent followers of Weiser's vision, I argue that it (and other agendas) will turn out to be more beneficial for society than persisting with following an unrealistic goal. Current technological developments together with emerging findings from user studies, showing how human activities have been positively extended by 'bounded' (as opposed to pervasive) technologies, suggest that much can be gained from re-conceptualizing UbiComp in terms of designing user experiences that creatively, excitedly, and constructively extend what people currently do. This does not mean that the main tenet of Weiser's vision be discarded (i.e., computers appearing when needed and disappearing when not) but rather we begin to entertain other possibilities – besides calmness – for steering UbiComp research. Examples include extending and supporting personal, cognitive and social processes such as habit-changing, problem-solving, creating, analyzing, learning or performing a skill. Ultimately, research and development should be driven by a better understanding of human activity rather than

what has tended to happen, namely, "daring to intervene, clumsily, in situations that already work reasonably well" [20, p231].

In the remainder of this paper I offer a constructive critique of Weiser's vision and the subsequent research that has followed in its footsteps. I then outline an alternative agenda for UbiComp, highlighting pertinent questions, concerns and illustrative examples of how it can be achieved.

2 Weiser's Vision Revisited and Early Research

To illustrate how his early vision of ubiquitous computing could work, Weiser [47] presented a detailed scenario about a day in the life of Sal, an executive single mother. The scenario describes what Sal gets up to, as she moves from her domestic world to her work place, during which she is perpetually informed of the goings on of her family, neighbors, fellow citizens and work colleagues. With this knowledge she is able to keep up-to-date, avoid obstacles, make the most of her time and conduct her work – all in smooth and effective ways. The scenario emphasizes coziness, comfort and effortlessness:

"Sal awakens: she smells coffee. A few minutes ago her alarm clock, alerted by her restless rolling before waking, had quietly asked "coffee?", and she had mumbled "yes." "Yes" and "no" are the only words it knows.

Sal looks out her windows at her neighborhood. Sunlight and a fence are visible through one, but through others she sees electronic trails that have been kept for her of neighbors' coming and going during the early morning. Privacy conventions and practical data rates prevent displaying video footage, but time markers electronic

tracks on the neighborhood map let Sal feel cozy in her street."

In this small excerpt we see how the world evolves around Sal's assumed needs, where computers, cameras and sensors are embedded into her world to make her life super efficient, smooth and calm. It is as if she glides through life, where everything is done or laid out for her and whenever there is potential for frustration, such as a traffic jam or parking problem, the invisible computers come to her rescue and gently inform her of what to do and where to go. It is worth drawing an analogy here with the world of the landed aristocracy in Victorian England who's day-to-day live was supported by a raft of servants that were deemed to be invisible to them. This scenario also highlights the ethical issues that such an informed world needs to address, namely the importance of establishing appropriate levels of privacy that are considered acceptable by a community (e.g., having abstract digital trails rather than video footage to ensure anonymity).

The core topics raised in Weiser's seminal papers have motivated much subsequent UbiComp research. Most prominent themes are context-aware computing, ambient/ubiquitous intelligence and recording/tracking and monitoring. (N.B. It should be noted that these are not mutually exclusive but overlap in the aims and methods used.)

2.1 Context-Aware Computing

Context-aware computing focuses on detecting, identifying and locating people's movements, routines or actions with a view to using this information to provide

relevant information that may augment or assist a person or persons. Many projects have been conducted under this heading to the extent that it has been noted that ubiquitous computing is sometimes called context-aware computing [12]. In a nutshell, context is viewed as something that can be sensed and measured using location, time, person, activity type and other dimensions. An example of an early context-sensitive application was comMotion that used location information and a speech output system to inform people when they were driving or cycling past a store to buy the groceries they needed [30].

A motivation behind much context-aware computing is to find ways of compensating for limitations in human cognition, e.g., attention, memory, learning, comprehension, and decision-making, through the use of sensor-based and computational tools. For example, augmented cognition – originating in military research – seeks to develop methods "to open bottlenecks and address the biases and deficits in human cognition" by continually sensing the ongoing context and inferring what strategies to employ to help people in their tasks [5].

Key questions in context-aware computing concern what to sense, what form and what kind of information to represent to augment ongoing activities. A number of location and tagging technologies have been developed, such as RFID, satellite, GPS and ultrasonics, to enable certain categories of information to be tracked and detected.

Many of these, however, have been beset with detection and precision limitations, sometimes resulting in unreliable and inaccurate data. Recent advances in cognitive radio technology that is software defined (SDR), promises to be more powerful; wireless systems will be able to locate and link to locally unused radio frequency, based on the ability to sense and remember various factors, such as human behavior, making them more dependable and more aware of their surroundings [4]. The advocates of this new technology portray its potential for highly complex settings, such as combat war zones to help commanders from different friendly forces stay appraised of the latest situation, through voice, data and video links, thereby reducing collateral damage [4].

While newer technological developments may enable more accurate data to be detected and collected it is questionable as to how effectively it can be used. It still involves Herculean efforts to understand, interpret and act upon in real-time and in meaningful ways. Context-aware systems that attempt to guide a person through certain activities require models of human behavior and intentionality that are based on rationality and predictability [40]. However, as already mentioned, people often behave in unpredictable and subtle ways in their day-to-day contexts. Therefore, it is likely that context-aware systems will only ever be successful in highly constrained settings.

2.2 Ambient and Ubiquitous Intelligence

Another dominant theme that has emerged in the field of UbiComp is ubiquitous or ambient intelligence, i.e., computational intelligence that is part of both the physical and the digital worlds. This approach follows on from work in artificial intelligence. The phrase ‘right place/right time/right means’ has been sloganized with visions of smart worlds and smart things, embedded with intelligence, that will predict people’s needs and react accordingly [25]. Instead of reaching for the remote to change the TV

channel the smart entertainment system will do it for us, instead of browsing the web the smart internet will find the information we need and so on. Just as it is becoming increasingly common place for supermarkets to automatically open their doors as we walk towards them, toilets to flush when we stand up and taps to release water as we wave our hands under them it is envisioned that information will appear on our TVs, watches, walls, and other displays as and when needed (e.g., children will be alerted of dangers and tourists will be informed of points of interest when walking through an unfamiliar city).

However, similar to context-aware computing, ambient intelligence is proving to be a hard nut to crack. While there have been significant advances in computer vision, speech recognition and gesture-based detection, the reality of multimodal interfaces – that can predict and deliver with accuracy and sensitivity what is assumed people want or need – is a long way off. One of the most well known attempts at implement-

ing ambient intelligence was IBM's BlueEyes project, that sought to develop computers that could "see" and "feel" like humans. Sensing technology was used to identify a person's actions and to extract key information that was then analyzed to determine the person's physical, emotional, or informational state. This was intended to be used to help make people "more productive by performing expected actions or by providing expected information." The success of the BlueEyes project, however, was limited; an example of an achievement that is posted on its website is of a television that would turn itself on when a person in the room made eye contact with it. To turn it off, the person could 'tell' it to switch off.

Such meager accomplishments in both context-aware computing and ambient intelligence reflect just how difficult it can be to get a machine to behave like a human. But it is essential that such systems be accurate for them to be accepted by humans in their everyday context. Reading, interpreting and acting upon people's moods, intentions, desires, etc, at any given moment in an appropriate way is a highly developed human skill that when humans get it wrong can lead to misunderstanding. When a ubiquitous computing system gets it wrong – which is likely to be considerably more frequent – it is likely to be more frustrating and we are likely to be less forgiving. For example, when the system decides to switch on the TV because we happen momentarily to stare into space while reading a book, it is likely to be unnerving and extremely annoying, especially if 'it' persistently gets it wrong.

2.3 Recording, Tracking and Monitoring

The push towards developing assistive applications through sensing and alerting has been most marked for vulnerable people; a number of UbiComp systems have been built to constantly check up on the elderly, the physically and mentally disabled [34]. The movements, habits, health and mishaps of such people are recorded, tracked and presented via remote monitors to the families, carers and other people responsible for them, who can then use the information to make decisions about whether to intervene or administer alternative forms of medical care or help. In particular, there has been a move towards developing ubiquitous computing systems to aid elderly people, who need to be cared for, by helping them take their medicines regularly, checking up on their physical health, monitoring their whereabouts and detecting when they have fallen over [e.g., 13].

A number of assisted living applications and services has also been developed to help people with loss of vision or deteriorating memory to be more independent in their lives. For example, Cyber Crumbs was designed to help people with progressive vision loss find their way around a building using a reader badge system that reads out directions and warns of obstacles, such as fire hydrants [39]. Cook's Collage was developed as an aid for people with memory loss. It replays a series of digital still images in a comic strip reel format depicting people's cooking actions in situ, in-

tended to help them remember if they have forgotten a step (e.g., adding a particular ingredient) after being distracted [45].

A reason for there being so much interest in helping the less able in UbiComp is that explicit needs and benefits can be readily identified for these user groups. Moreover, there is an assumption that pervasive technologies offer more flexibility and scope for providing solutions compared with other computing technologies since they can sense, monitor and detect people's movements, bodily functions, etc., in ways not possible before. There is a danger, however, that such techniques may probe too far into the lives of less able people resulting in – albeit unintentionally – 'extreme' forms of recording, tracking and monitoring that these people may have no control over. For example, consider the extent to which a group of researchers went to in order to help with the care of old people in a residential care home [6]. A variety of monitoring devices were installed in the home, including badges on the patients and the caregivers and switches on the room doors that detected when they were open or closed. Load sensors were also used to measure and monitor weight changes of people while in their beds; the primary aim was to track trends in weight gain or loss over time. But the sensors could also be used to infer how well someone was sleeping. If significant movement was detected during the night this could enable a caregiver to see whether the person was having trouble sleeping (and if there was a huge increase in weight this could be inferred as someone else getting in or on the bed).

Such panopticon developments elicit a knee-jerk reaction of horror in us. While the motives behind such projects are altruistic they can also be naïve, overlooking how vulnerable people's privacy and self-respect may be being violated. Not surprisingly, there has been enormous concern by the media and other social scientists about the social implications of recording, tracking and re-representing people's movements, conversations, actions and transactions. Inevitably, a focus has been on the negative aspects, namely a person's right to privacy being breached. Is it right to be videoing and sensing people when sleeping, eating, etc., especially when they are not at their best [2]? Is it right to be providing information to other family members about their granny's sleeping habits, especially if it can be inferred from the sensed data that she might have got into bed with another patient, which none of the vested parties might want to share or let the others know about.

While most projects are sensitive to the privacy and ethical problems surrounding the monitoring of people, they are not easy to solve and have ended up overwhelming UbiComp research. Indeed, much of the discussion about the human aspects in the field has been primarily about the trade-offs between security and privacy, convenience and privacy, and informedness and privacy. This focus has often been at the expense of other human concerns receiving less airing, such as how recording, tracking and re-representing movements and other information can be used to facilitate social and cognitive processes.

My intention here is not to diminish the importance of awareness, ambience and monitoring to detect and inform people in their everyday lives, together with the ethical and social issues they raise. Rather, my overview of the projects in these areas has revealed how difficult it is to build calm computing systems and yet the attempts have largely dominated the field of UbiComp. Those that have tried have fallen short, resulting in prototype systems that can sometimes appear to be trivial or demeaning. Conversely, there has been less focus on other areas of research that could prove to be easier to achieve and potentially of more benefit to society. The time is ripe for other directions to take center stage in UbiComp. One such avenue promoted here is to consider how humankind's evolved practices of science, learning, health, work and play can be enhanced. This involves thinking about UbiComp not in terms of embedding the environment with all manner of pervasive technologies but instead as bounded ensembles of entities (e.g., tools, surfaces and lenses) that can be mobile, collaborative or remote, through which information, other people and the environment are viewed and interacted with when needed. Importantly, it argues for rethinking the nature of our relationship with the computer.

3 A New Agenda for UbiComp: Engaging User Experiences

I suggest here that it is highly profitable to recast UbiComp research in the context of a central motivation that computers were originally designed for, namely, as tools, devices and systems that can extend and engage people in their activities and pursuits. My reason for proposing this is based on the success of researchers who have started to take this approach. In particular, a number of user studies, exploring how UbiComp technologies are being appropriated, are revealing how the 'excitement of interaction' can be brought back in innovative ways; that is not frustrating and which is quite different from that experienced with desktop applications. For example, various mixed reality, physical-digital spaces and sensor-rich physical environments have been developed to enable people to engage and use multiple dynamic representations in novel ways: in scientific and working practices and in collaborative learning and experimental games. More extensive inquiries and decisions have been enabled in situ, e.g., determining the effects of deforestation in different continents and working out when is the best time to spray or pick grapes in a vineyard.

Recently, world famous computer scientist John Seely Brown put forward his updated vision of UbiComp 1 in a keynote, outlining 'a common sense' model that emphasizes how UbiComp can help to catalyze creativity [41]. He proposed that creating and learning be seen as integral to our work and leisure that are formed through recreation and appropriation activities. In a similar vein, I argue that it is timely to switch from a reactive view of people towards a more proactive one. Instead of augmenting the environment to reduce the need for humans to think for themselves about what to do, what to select, etc., and doing it for them, we should consider how Ubi-

Comp technologies can be designed to augment the human intellect so that people can perform ever greater feats, extending their ability to learn, make decisions, reason, create, solve complex problems and generate innovative ideas. Weiser's idea that

1 John Seely Brown was a co-author of the paper written by Weiser on calm technology.

technologies be designed to be 'so embedded, so fitting and so natural' that we use them without thinking about them needs to be counter-balanced; we should also be designing them to be exciting, stimulating and even provocative – causing us to reflect upon and think about our interactions with them. While Weiser promoted the advantages of calm computing I advocate the benefits of engaging UbiComp experiences that provoke us to learn, understand and reflect more upon our interactions with technologies and each other.

A central concern of the engaging UbiComp experiences agenda is to fathom out how best to represent and present information that is accessible via different surfaces, devices and tools for the activity at hand. This requires determining how to make intelligible, usable and useful, the recordings of science, medicine, etc., that are streaming from an increasing array of sensors placed throughout the world. It also entails figuring out how to integrate and replay, in meaningful and powerful ways, the masses of digital recordings that are begin gathered and archived such that professionals and researchers can perform new forms of computation and problem-solving, leading to novel insights. In addition, it involves experimenting more with creative and constructive uses of UbiComp technologies and archived digital material that will excite and even make people feel uncomfortable.

In terms of who should benefit, it is useful to think of how UbiComp technologies can be developed not for the Sal's of the world, but for particular domains that can be set up and customized by an individual firm or organization, such as for agriculture production, environmental restoration or retailing. At a smaller scale, it is important to consider how suitable combinations of sensors, mobile devices, shared displays, and computational devices can be assembled by non-UbiComp experts (such as scientists, teachers, doctors) that they can learn, customize and 'mash' (i.e., combine together different components to create a new use). Such toolkits should not need an army of computer scientists to set up and maintain, rather the inhabitants of ubiquitous worlds should be able to take an active part in controlling their set up, evolution and destruction. Their benefits should be clear: enabling quite different forms of information flow (i.e., ways and means of accessing information) and information management (i.e., ways of storing, recording, and re-using information) from older technologies, making it possible for non-UbiCompers to begin to see how to and subsequently develop their own systems that can make a difference to their worlds. In so doing, there should be an emphasis on providing the means by which to augment and extend existing practices of working, learning and science.

As quoted by Bruner [10] “to assist the development of the powers of the mind is to provide amplification systems to which human beings, equipped with appropriate skills, can link themselves” (p.53). To enable this to happen requires a better understanding of existing human practices, be it learning, working, communicating, etc. Part of this reconceptualization should be to examine the interplay between technologies and their settings in terms of practice and appropriation [15]. “Practices develop around technologies, and technologies are adapted and incorporated into practices.” (Dourish, 2001, p. 204). More studies are needed that examine what people do with their current tools and devices in their surrounding environments. In addition, more studies are needed of UbiComp technologies being used in situ or the wild – to help illuminate how people can construct, appropriate and use them [e.g., 16, 22, 23, 29].
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With respect to interaction design issues, we need to consider how to represent and present data and information that will enable people to more extensively compute, analyze, integrate, inquire and make decisions; how to design appropriate kinds of interfaces and interaction styles for combinations of devices, displays and tools; and how to provide transparent systems that people can understand sufficiently to know how to control and interact with them. We also need to find ways of enabling professionals and laypeople alike to build, adapt and leverage UbiComp technologies in ways that extend and map onto their activities and identified needs.
A more engaging and bounded approach to UbiComp is beginning to happen but in a scattered way. Three of the most promising areas are described below: (i) playful and learning practices, (ii) scientific practices and (iii) persuasive practices. They show how UbiComp technologies can be developed to extend or change human activities together with the pertinent issues that need to be addressed. Quite different practices are covered, reflecting how the scope of UbiComp can be broad but at the same time targeted at specific users and uses.
3.1 Playful and Learning Practices
One promising approach is to develop small-scale toolkits and sandboxes, comprising interlinked tools, digital representations and physical artifacts that offer the means by which to facilitate creative authoring, designing, learning, thinking and playing. By a sandbox it is not meant the various senses it has been used in computing but more literally as a physical-digital place, kitted out with objects and tangibles to play and interact with. Importantly, these should allow different groups of people to participate in novel activities that will provoke and extend existing repertoires of technology-augmented learning, playing, improvising and creating. An example of a promising UbiComp technology toolkit is PicoCrickets, developed at MIT Media Lab, arising from the work of Mitch Resnick and his colleagues. The toolkit comprises sensors, motors, lights, microcomputers, and other physical and electrical devices that can be easily programmed and assembled to make them react, interact and communicate,

enabling “musical sculptures, interactive jewelry, dancing creatures and other playful inventions” to be created by children and adults alike. An advantage of such lightweight, off-the-shelf tangible toolkits is that they offer many opportunities for different user groups (e.g., educators, consultants) to assemble and appropriate in a range of settings, such as schools, waiting rooms, playgrounds, national parks, and museums. A nagging question, however, is how do the benefits of such UbiComp toolkits and sand boxes compare with those offered by more conventional ones – that are much cheaper and more practical to make? Is it not the case that children can be highly creative and imaginative when given simply a cardboard box to play with? If so, why go to such lengths to provide them with new tools? The debate is redolent of whether it is better for children to read a book or watch a 3D Imax movie. One is not necessarily better than the other: the two provide quite different experiences, triggering different forms of imagination, enjoyment and reflection. Likewise, UbiComp and physical toys can both provoke and stimulate, but promote different kinds of learning and collaboration among children. However, a benefit of UbiComp toolkits over physical artifacts is that they offer new opportunities to combine physical interaction, through manipulation of objects or tools or through physical body postural movement and

location, with new ways of interacting, through digital technology. In particular, they provide different ways of thinking about the world than interacting solely with digital representations or solely with the physical world. In turn, this can encourage or even enhance further exploration, discovery, reflection and collaboration [35].

Examples of projects that have pioneered the design of novel physical-digital spaces to facilitate creativity and reflection include the Hunting of the Snark [32], Ambient Wood [36], RoomQuake [33] Savannah [17], Environmental Detectives [27], Drift Table [19] and Feeding Yoshi [7]. Each of these have experimented with the use of mobile, sensor and fixed technologies in combination with wireless infrastructures to encourage exploration, invention, and out of the box thinking.

The Hunting of the Snark adventure game provoked young children into observing, wondering, understanding, and integrating their fragmented experiences of novel physical-digital spaces that subsequently they reflected upon and shared as a narrative with each other. A combination of sensor-based, tangible, handheld and wireless technologies was used to create the physical-digital spaces, where an imaginary virtual creature was purported to be roaming around in. The children had to work out how to entice the creature to appear in them and then gather evidence about its personality, moods, etc, by walking with it, feeding it and flying with it. Similarly, Savannah was designed as a physical-digital game to encourage the development of children’s conceptual understanding of animal behavior and interactions in an imaginary virtual world. The project used GPS and handheld computers to digitally overlay a school playing field with a virtual plain. Children took on the roles of lions, had to

hunt animals in the virtual savannah and capture them to maintain energy levels. After the game, the children reflected on their experiences by interacting with a visualization on a large interactive whiteboard, that showed the trails they made in the Savannah and the sounds and images that they encountered at specific place.

The Ambient Wood project used an assortment of UbiComp technologies to encourage more self-initiation in inquiry and reflective learning. Various wireless and sensor technologies, devices and representational media were combined, designed and choreographed to appear and be used in an 'ambient' woodland. Several handcrafted listening, recording and viewing devices were created to present certain kinds of digital augmentations, such as sounds of biological processes, images of organisms, and video clips of life cycles. Some of these were triggered by the children's exploratory movements, others were collected by the children, while still others were aggregated and represented as composite information visualizations of their exploratory behavior.

RoomQuake was designed to encourage children to practice scientific investigatory practices: an earthquake was simulated in a classroom using a combination of interconnected ambient media, string and physical styrofoam balls. The ambient media provided dynamic readings of the simulated earthquakes, which students then represented as physical models using the physical artifacts. The combination of computer-based simulations and physical-based artifacts enabled the whole class to take part in the measuring, modeling, interpreting, sparking much debate and reflection among the children about the seismic events.

As part of the Equator collaboration, a number of innovative 'seamful games' have been developed. The inherent limitations of ubiquitous technologies have been deliberately exploited to provoke the players into thinking about and acting upon their significance to the ongoing activity. Two examples are Treasure in which players had

to move in and out of a wireless network connectivity to collect and then deposit gold tokens and Feeding Yoshi where the players were required to feed virtual creatures scattered around a city with virtual fruits that popped up on their displays as a result of their location and activity therein.

Evaluations of this emerging genre of physical-digital spaces for learning and playing have been positive, highlighting enhanced understanding and an immense sense of engagement. Children and adults have been able to step back and think about what they are doing when taking part in the game or learning experience, examining the rationale behind their choices when acting out and interacting with the UbiComp-based technologies in the space. However, many of the pioneering projects were technology, resource and researcher intensive. While guidance is now beginning to appear to help those wanting to design UbiComp-based learning and playing experiences [e.g., 9, 36] we need also to strive towards creating the next generation of physical-digital spaces and toolkits that will be as easy, cheap and popular to construct as Lego

kits once were.

3.2 Scientific Practices

Another area where UbiComp has great potential for augmenting human activities is the practice of scientific inquiry and research. Currently, the sciences are going through a major transformation in terms of how they are studied and the computational tools that are used and needed. Microsoft's 2020 Science report – a comprehensive vision of science for the next 14 years written by a group of internationally distinguished scientists – outlines this paradigm shift [31]. It points out how new conceptual and technological tools are needed that scientists from different fields can "understand and learn from each other's solutions, and ultimately for scientists to acquire a set of widely applicable complex problem solving capabilities". These include new programming, computational, analysis and publication tools. There is much scope, too, for utilizing UbiComp technologies to enhance computation thinking, through integrating sensor-based instrumentation in the medical, environmental and chemical sciences. The ability to deliver multiple streams of dynamic data to scientists, however, needs to be matched by powerful interfaces that allow them to manipulate and share them in new ways, from any location whether in the lab or in the field. Areas where there is likely to be obvious benefits to scientists through the integration of UbiComp and computational tools are environmental science and climate change. These involve collaborative visualization of scientific data, mobile access to data and capture of data from sensors deployed in the physical world. Being able to gain a bigger, better and more accurate picture of the environmental processes may help scientists make more accurate predictions and anticipate more effectively natural disasters, such as tsunamis, volcanoes, earthquakes and flooding. However, it may not simply be a case of more is more. New ways of managing the burgeoning datasets needs to be developed, that can be largely automated, but which also allows scientists to have effective windows, lenses etc., into so that they can interpret and make intelligible inferences from them at relevant times.

The 2020 report notes how tomorrow's scientists will need to make sense of the masses of data by becoming more computationally literate – in the sense of knowing how to make inferences from the emerging patterns and anomalies that the new

generation of software analysis tools provide. To this end, a quite different mindset is needed in schools for how science is taught. The design of new learning experiences that utilize UbiComp technologies, both indoors and outdoors, need to be developed to seed in young children the sense of what is involved in practicing new forms of complex, computational science. An example of how this can be achieved is the embedded phenomena approach; scientific phenomena are simulated using UbiComp technologies, for long periods of time, to create opportunities for groups of students to explore 'patient' science [32]. Essentially, this involves the accumulation, analysis

and representation of data collected from multiple computational devices over extended periods of observation in the classroom or other sites. In so doing, it allows students to engage in the collaborative practice of scientific investigation that requires hard computational thinking but which is also exciting, creative and authentic. A core challenge, therefore, is to find ways of designing novel science learning experiences that capitalize on the benefits of combining UbiComp and PC technologies that can be used over extended periods.

3.3 Persuasive Practices

The third area where there is much potential for using UbiComp technologies to engage people is as part of self-monitoring and behavioral change programs. While a range of persuasive technologies (e.g., adverts, websites, posters) has already been developed to change people's attitudes and behaviors, based on models of social learning [18], UbiComp technologies provide opportunities for new techniques. Specifically, mobile devices, such as PDAs coupled with on-body sensors, can be designed to enable people to take control and change their habits or lifestyles to be healthier by taking account of and acting upon dynamically updated information provided by them. For example, Intille and his group are exploring how mobile computational tools for assessing behavioral change, based on social psychology models, can be developed to motivate physical activity and healthy eating.

A key question that needs to be addressed is whether UbiComp technologies are more (or less) effective compared with other technologies in changing behavior. A diversity of media-based techniques (e.g., pop-up warning messages, reminders, prompts, personalized messages) has been previously used to draw people's attention to certain kinds of information to change what they do or think at a given point. In terms of helping people give up habits (e.g., smoking, excessive eating) they have had mixed results since people often relapse. It is in the long-term context that UbiComp technologies may prove to be most effective, being able to monitor certain aspects of people's behavior and represent this information at critically weak moments in a cajoling way. A constant but gentle 'nagging' mechanism may also be effective at persuading people to do something they might not have otherwise done or to not to do something they are tempted to do. For example, a collaborative cell phone application integrated with a pedometer was used to encourage cliques of teenage girls to monitor their levels of exercise and learn more about nutrition in the context of their everyday activities [44]. The software was designed to present the monitored process (e.g., walking) in a way that made it easy for the girls to compute and make inferences of how well they were doing in terms of the number of steps taken relative to each other. A preliminary study showed that such a collaborative self-monitoring system was

effective at increasing the girl's awareness of their diet, level of exercise and enabling them to understand the computations involved in burning food during different kinds

of exercise. But most significantly, it enabled the girls to share and discuss this information with each other in their private clique, capitalizing on both the persuasive technology and peer pressure.

Incorporating fun into the interface can also be an effective strategy; for example, Nintendo's Pocket Pikachu with pedometer attached was designed to motivate children into being more physically active on a consistent basis. The owner of the digital pet that 'lives' in the device is required to walk, run or jump each day to keep it alive. If the owner does not exercise for a week the virtual pet becomes unhappy and eventually dies. This can be a powerful means of persuasion given that children often become emotionally attached to their virtual pets, especially when they start to care for them.

UbiComp technologies can also be used to reduce bad habits through explicitly providing dynamic information that someone would not have been aware of otherwise. In so doing, it can make them actively think about their behavior and modify it accordingly. The WaterBot system was developed using a special monitoring and feedback device to reduce householder's usage of water in their homes – based on the premise that many people are simply unaware of how wasteful they are [3]. A sensor-based system was developed that provided positive auditory messages and chimes when the tap was turned off. A central idea was to encourage members of the household to talk to one another about their relative levels of water usage provided by the display and to try to out do one another in the amount of water used.

But to what extent do UbiComp technologies, designed for persuasive uses, differ from the other forms of monitoring that were critiqued earlier in the paper? A main difference is that there is more active involvement of those being monitored in attaining their desired behavior change compared with those who were being monitored and assisted in care homes. The objective is to enable people, themselves, to engage with the collected information, by monitoring, understanding, interpreting and acting upon it – and not the environment or others to act upon their behalf. Much of the research to date in UbiComp and healthcare has focussed on automated bio-monitoring of physiological processes, such as EEGs and heart rate, which others, i.e., specialists, examine and use to monitor their patient's health. In contrast, persuasive technologies are intended to provide dynamic information about a behavioral process that will encourage people from doing or not doing something, by being alerted and/or made aware of the consequences of what they are about to do. Moreover, designing a device to be solely in the control of the users (and their social group) enables them to be the owners of the collected data. This circumvents the need to be centrally concerned with privacy issues, allowing the focus of the research to be more oriented towards considering how best to design dynamically updated information to support cognitive and social change. A challenge, however, in this area is for long term studies to be conducted that can convincingly show that it is the perpetual and

time-sensitive nature of the sensed data and the type of feedback provided that contributes to behavioral modification.

4 Conclusions

Many of the research projects that have followed in the footsteps of Weiser's vision of calm computing have been disappointing; their achievements being limited by the extent to which they have been able to program computers to act on behalf of humans. Just as 'strong' AI failed to achieve its goals – where it was assumed that "the computer is not merely a tool in the study of the mind; rather, the appropriately programmed computer really is a mind" [41], it appears that 'strong' UbiComp is suffering from the same fate. And just as 'weak' AI 2 revived AI's fortunes, so, too, can 'weak' UbiComp bring success to the field. This will involve pursuing more practical goals and addressing less ambitious challenges; where ensembles of technologies are designed for specific activities to be used by people in bounded locations. To make this happen, however, requires moving from a mindset that wants to make the environment smart and proactive to one that enables people, themselves, to be smarter and proactive in their everyday and working practices. Three areas of research were suggested as to how this could be achieved; but, equally, there are others where there is much potential for enhancing and extending human activities (e.g., vineyard computing [11], firefighting [24] and sports). As part of the expansion of UbiComp, a wider range of human aspects should be considered, drawing upon alternative theory, guiding frameworks and metaphors [c.f. 8, 15]. To enable other human concerns to become more prominent, however, requires the hefty weight of privacy and other related ethical issues on UbiComp's shoulders to be lessoned.

The 'excitement of interaction' that Weiser suggested forsaking in the pursuit of a vision of calm living should be embraced again, enabling users, designers and researchers to participate in the creation of a new generation of user experiences that go beyond what is currently possible with our existing bricolage of tools and media. We should be provoking people in their scientific, learning, analytic, creative, playing and personal activities and pursuit. Finally, while we have been privileged to have had such a great visionary, whose legacy has done so much to help shape the field, it is timely for a new set of ideas, challenges and goals to come to the fore and open up the field.

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The Lucent Web site is built hierarchically, in the sense that pages deeper in the directory tree represent more detailed information than those at shallower levels. At its busiest, there can be as many as 300 people browsing www.lucent.com; while during the pre-dawn hours there can be as few as 5 simultaneous visitors. Our sonification is designed to convey qualitative information about site usage, answering questions like:

Overall, is the site busy or quiet?

What proportion of the visitors are delving for specific information deep within the site, as compared to those visitors who are "just passing through," glancing briefly at the home page and then moving on?

How are users distributed across the various content areas of the site?

Which portions of the site are visited together? What kinds of patterns do we find in user behavior?

We think of this sonification as one possible "background" information stream that can inform content providers, Web designers and even the visitors themselves.

2.1.1. Sonification design

Our audio display makes use of the hierarchical structure of the content offered by www.lucent.com. First, a unique pitch was used to identify each of five high-level subdomains within the site: /micro, representing Lucent's microelectronics design and manufacturing business (now Agere Systems); /enterprise, for the enterprise systems and software business (now Avaya Communications); /minds, a corporate introduction to Bell Labs research; /press, a collection of press releases and investor information; and /search, the local search engine for the site.

The total number of visitors accessing any information from a subdomain affects the loudness and tonal balance of a low-register drone at the associated pitch. Visitors requesting content deeper in the site are represented by higher-pitched pulsing tones (separated by one or two octaves from the base pitch for the subdomain):

the faster the pulse, the more people are accessing that area, and the greater the proportion of high-register sounds, the more detailed the content. By assigning well-separated pitches to each subdomain, shifts in activity both within and between the areas can be heard. In Table 1 we present a simple mapping of data collected by the Lucent Web server to a continuously time-varying vector of usage statistics. In the category of Overall browsing, we count any visitor accessing content pages (HTML, PostScript or PDF) from the indicated subdomain. A Mid-Level access is a request for content two or more directories down. Simple examples are /micro/K56flex/index.html (information on a brand of 56K modem) and /press/0101/010118.nsb.html (a press release for January 18, 2001). The final category, Deep browsing, refers to pages that are four or more directories down in the tree. One example is a paper from the April/June 2000 issue of the Bell Labs Technical Journal, located at /minds/techjournal/apr-jun2000/pdf/paper02.pdf.

Then, the resulting 15 values in Table 1, A1–E3, were mapped to sound as follows:

Overall activity Measured by A1–E1, voiced with a low-register drone. The aggregate number of visitors accessing information within each of the five areas modulates the loudness of each of the five pitches.

/micro /enterprise /minds /press /search
Overall A1 B1 C1 D1 E1
Mid-Level A2 B2 C2 D2 E2
Deep A3 B3 C3 D3 E3

Table 1: Mapping used for Web site traffic example. Overall activity records the movements of all users; Mid-Level counts users 2 or 3 directories into the site; Deep browsing consists of users 4+ directories down.

Mid-Level browsing Measured by A2–E2 and assigned a rhythmic middle-register tone pulse; pulse loudness and repetition speed rises and the timbral brightness increases as the volume of mid-level browsing increases. There are five independent pulses, each at a different fixed pitch, representing the five content areas.

Deep browsing Measured by A3–E3 and made audible via rhythmic high-register “ting” sounds (plucked steel string samples). Loudness and repetition speed rises as the volume of

deep browsing increases. Again, there are five independent "ting" sounds, each at a different fixed pitch, representing the five content areas.

We used pitch groups that were consonant, and for the sounds that incorporated rhythm (A2–E3), the phase and frequency of each pulse in the matrix varies independently, yielding a sound with a changing rhythmic texture but no fixed beat.

The purpose of this sonification is to make interpretable the activities of users on a Web site. Therefore, the stream of hits being processed by a Web server (reduced to include only the HTML, PostScript and PDF documents) needs to be transformed to extract meaningful user-level data. A real-time monitoring tool was developed that maintains a bank of active visits (recording separately the activities of all the people browsing the site at a given time) and updates various statistics with each user request. When cookies or some other authentication mechanism allows us to recognize returning visitors, the monitor will update a more complicated user profile that encapsulates previous browsing patterns. Our traffic sonification as described above takes as input the location of each visitor within a site at a given point in time. When constructing more elaborate sound displays, our design will continue to focus on user activities, drawing more heavily on the statistics culled by the monitoring tool. This emphasis distinguishes our approach from sonification methods that assess Web server performance by making audible statistics relating to server load, HTTP errors, and agent types [?].

2.1.2. Impressions and extensions

We have created three audio examples for the activity on the Lucent site. Our data were captured on November 11, 1999 and we created sonifications of the traffic at 6:00 am, an extremely slow period for the site; noon, a relatively active time; and 2:30 pm, the point at which the site was busiest. The samples are located at our project Web site [6]. Even with this relatively straightforward mapping, one finds compelling patterns. For example, the affinity between the /enterprise subdomain and the /search facility can be heard as the pulses for these areas rise and fall together.[3]

[3] While clearly audible, these shifts can really only be precisely associated with areas after a certain amount of experience with the mapping.

Also, when comparing moderately active to extremely busy periods, we find that the number of people digging deep into the site is not a fixed fraction of the total number of visitors. That is, the volume of the low-register drones exhibits much more variation than the components for the other two categories of accesses. Each of these effects can be verified by examining the logs, reinforcing the usefulness of our sonification as a tool for constructing hypotheses about site traffic.

As mentioned at the beginning of this section, Web browsers offer a rich set of data about the visitor when requesting data from a server. This display makes use of only the most basic information about a visit, namely the depth of pages accessed. In ongoing work, we are augmenting our sonification with extra features derived both directly from the server data as well as from statistical navigation models [12] fit for the Web site under study. So far, we have found that such extensions are most effective when developed in the context of a particular monitoring application. For example, an extended version of this ambient display can aid system architects of large, Web hosting services understand cache performance and can aid in server provisioning. Another extension will make greater user of our navigation models and can help designers and usability engineers better architect Web sites. We will report on these and other developments through the project Web site [4].

2.2. Chat rooms and bulletin boards

At any given moment, tens of thousands of real-time conversations are taking place across the Internet on public forums, bulletin boards and chat sites. To imagine making these conversations simultaneously audible evokes an image of uproarious babble. And yet, in the aggregate, this massive stream of live communication could exhibit rich thematic structure. Can we find a meaningful way to listen in to so many conversations, rendering them in a way that is comprehensible and not overwhelming?

In some sense, a byproduct of our Web traffic sonification is the creation of a kind of community from the informal gathering of thousands of visitors to a given Web site. Traditionally, informational Web sites like www.lucent.com have provided us with very little sense of the other people who are requesting data from the server. To attract and retain visitors, however, many commercial sites recognize the potential of the Web to form so-

cial as well as informational networks. As a result, Web-based forums, message boards and a variety of chat services are common components of current site designs. While Internet Relay Chat (IRC) has been a widely used standard since the inception of the Internet, the popularization of the Web has resulted in a virtual explosion of chat applications.4 For example, www.yahoo.com (a US-based Web portal) offers hundreds of separate chat rooms attracting tens of thousands of visitors a day. Specialized sites like www.style.com (the homepage for Vogue magazine) or www.audiworld.com (an resource for Audi owners) have also found their message boards to be the most frequently accessed parts of their domains.

To get a sense of the amount of content that is available in these dynamic formats, we examined sites contained in the DMOZ Open Directory [3], an open source listing of over 2 million Web sites compiled and categorized by 33,000 volunteer editors. From the November 20, 2000 image of the directory, we counted 36,681 separate sites offering some kind of chat, bulletin board or other public forum. While we did not examine the activity on all of these sites, the number is staggering. If we include other peer-to-peer communication technologies like instant messaging,5 the amount of dialogue taking place on the Web at any point in time is almost unfathomable. The goal of our second sonification is to make interpretable the thousands of streams of dynamic information being generated on the Web. In so doing, we attempt to characterize a global dialogue, integrating political debates, discussions of current events, and casual exchanges between members of virtual communities.

4 RC was developed by Jarkko Oikarinen in Finland in the late eighties, and was originally intended to work as a better substitute for talk on his bulletin board.

2.2.1. Content monitors and the statistics engine

Our starting point is text. Albeit diverse in style and dynamic in character, the text (or transcript) of these data sources carries their meaning. Therefore, any auditory display consisting only of generated tones would not be able to adequately represent the data without a very complex codebook. The design of our sonification then depends heavily on text-to-speech (TTS). As with the traffic example in the previous section, we think of the audio output as another background information stream. The incorporation

of spoken components in the sound design poses new challenges, both practical and aesthetic. For example, simply voicing every word taking place in a single chat room can produce too much text to be intelligible when played in real-time and can quickly exhaust the listener. Instead, we build a hierarchical representation of the text streams that relies on statistical processing for content organization and summarization prior to display.

Before considering sonification design, we first had to create specialized software agents that would both discover new chat rooms and message boards, as well as harvest the content posted to these sites. (See Figure 1 for an overview of our system architecture.) Most bulletin boards and some chat applications use standard HTML to store visitor contributions. In many cases, a specific login name is required to gain access to the site. For these situations, we constructed a content agent in Perl, as this language provides us the most convenient platform for managing access details (like cookies). The public chat rooms on sites like chat.yahoo.com can be monitored in this way. For IRC we built a configurable Java client that polls a particular server for active channels. Web sites like www.cnn.com (a popular news portal) and www.financialchat.com (a financial community hosting chat services for day traders) offer several IRC rooms, some of which are tightly moderated.

In addition to collecting content, each monitoring agent also summarizes the chat stream, identifying basic topics and updating statistics about the characteristics of the discussion: What percentage of visitors are contributing? How often to they contribute and at what length? Is the room "on topic," or are many visitors posting comments on very different subjects? Topics are derived from the chat stream using a variant of generalized sequence mining [7] that incorporates tags for the different parts of speech. While the exact details are beyond the scope of this abstract, a generalized sequence is a string of words possibly separated by a wildcard, "*". For example, if we let A, B and C denote specific "contentful" words (say, nouns, adjectives and adverbs), then AB C , A B C and A B C are all generalized sequences. The wildcard allows us to identify "Gore * disputes * election" from the sentences

5 AOL alone records tens of millions of people using their instant messaging service each month.

Figure 1: System architecture overview. A large number of content streams (Chat = chat rooms; BB = Bulletin boards) are gathered by specialized agents that transmit them in a homogenized format to the statistics engine. The statistics engine then distills the streams into a much smaller number of configurable text streams as well as a number of descriptive vectors. The sonification engine then "plays" these text and data streams. The entire systems operates in real-time.

"Vice President Gore filed papers to dispute the presidential election," "Aides for Gore indicated that he has every reason to dispute the election", and "Gore is still deciding whether or not to dispute the election".

As many posts to chat rooms contain spelling mistakes and

incorrect grammar, assigning words to different parts of speech is error-prone. However, unlike most applications of statistical natural language processing, our content monitors update their summaries each time new material is posted and downweight older contributions. Because our sonification renders these sources in real-time, small mistakes have little effect on the power of the overall display to convey the ideas being discussed.

Each of the content monitors are periodically polled by the statistics engine (see Figure 1). This Java-application clusters the different chat rooms and bulletin boards based on their topic and numerical summaries. As the topic in a room changes over time, the statistics engine is constantly updating and reformulating cluster membership. Because a content stream can in fact support a number of simultaneous discussions (the threads of a bulletin board, say), we employ a soft-clustering technique. In our initial work, we have used a mixture-based scheme that determines the number of clusters with an MDL (Minimum Description Length) criterion [9]. Each room is then assigned a probability that it belongs to the different groups. This model also provides for topic summarization at the cluster-level. Next, a stochastic framework was developed to sample representative sentences posted to the chat or bulletin board. When a discussion is extremely unstructured, this selection is essentially random sampling from all the contributions added to the chat since the last polling point. In addition to textual data streams, the statistics engine is also responsible for communicating the various ingredients for the display to our sonification engine, Max/MSP [2] (see Figure 1). We have adopted the Open Sound Control [13] protocol from Center for New Music and Audio Technologies to transfer data between the statistics engine (running on a Macintosh with LinuxPPC) and the sonification engine (running on a Macintosh with OS/9).

2.2.2. Sonification design

As with the previous example (Section 2), our goal is to create a sonification that is both communicative and listenable. Here we face the additional challenge of incorporating verbal content. With TTS annotations, it becomes more difficult to intelligibly convey more than one layer of information through the audio channel. Our design incorporates spatialization, pitch and timbral differentiation, and rhythm to achieve clarity in the presentation of the hierarchically structured data coming from the statistics engine.

The auditory display cycles through topic clusters, spending relatively more time on subjects being actively discussed by the largest numbers of people. Each different topic is assigned a different pitch group, reinforcing subject changes when they occur. For each cluster, the statistics engine sends three streams of information to the sonification engine:

Topics A continuously updated list of up to ten "topics" (the most frequently appearing words and phrases – generalized sequences – mined from the multiple chat streams associated with the given cluster; the number of topics is configurable, but ten was chosen based on timing considerations);

Content samples A selection of sample sentences, identified by the statistics engine as typical or representative, in which these topics appear;

Content entropy A vector that represents the changing level of entropy in the source data.

The topics are spoken by the TTS system6 at regular intervals in a pitched monotone, and are panned alternately hard left and hard right in the stereo field, creating a sort of rhythmic "call and response." The sample sentences are panned center, and rendered with limited inflection (as opposed to the pitched monotone of the topics). The tonal, rhythmic and spatial qualities of the topics contrasts sufficiently with the sample sentences to create two distinctly comprehensible streams of verbal information.

The entropy vector controls an algorithmic piano score. When entropy is minimal and the discussion in the chat room or bulletin board is very focused on one subject, chords are played rhythmically in time with the rhythmic recitation of the topics. As entropy increases and the conversations diverge, a Gaussian distribution is used to expand the number, range and dynamics of notes that fall between the chords. With this audio component, one can easily differentiate a well-moderated content source from a more free-form, public chat without distracting from the TTS annotations. The piano score also serves a secondary function as an accompaniment to the vocal foreground, enhancing the compositional balance and overall musicality of the sound design.

2.2.3. Sample sonification and impressions

On our project Web site [5], we have a sample chat room sonification that cycles through three topics. In this sound file, we are

6 The built-in MacOS TTS capability controlled by Max/MSP.

listening to the output of only three content monitors. Hence, by design, each topic is confined to a single site. The first portion of this example (ending at 1:47 into the sample) concerns the recent recall of Bridgestone tires and was based on a www.cnn.com chat room. This discussion was heavily moderated and hence the backing piano score frequently reduces to a simple rhythm. For our second topic (from 1:47 to 3:21 of the sample) we recorded chat exchanges on www.financialchat.com one morning when Yahoo's stock opened low. In this example, we hear day traders frantically exchanging predictions about when Yahoo's stock will "bounce." The final topic in this sample (from 3:21 to the end) is again from www.cnn.com and treats a recent strike by the Screen Actor's Guild and the American Federation of Television and Radio Artists. This chat room was much less moderated than the previous CNN chat, and the backing piano score reflects that. Although this example does not make full use of the clustering capabilities of the statistics engine, the essence of our sonification design is clear. The audio display provides an informative and accessible representation of dynamic, textual content. The topic and content sample streams are easy to separate, and when placed in the background, call our attention to important new subjects being discussed on the Web.

2.2.4. Applications and Extensions

Our sonification provides an audible interface to the (now) massive amount of dynamic content available on the Web. Given the preprocessing that takes place in the content monitors and the statistics engine, a simple extension is to provide search-like functionality. A user can register interest in a certain topic and "tune" our display to present only rooms where this subject is being discussed. The necessary ingredients to implement this feature are all currently available in the statistics engine. Similarly, one can easily restrict the sites that are used for the display. When a new subject appears that draws the user's interest, it is also trivial to add a feature that would direct the user's browser to one or more chats associated with the topic. As a final extension, we have provided the content monitors with a configurable list of Web sites that can be used to help disambiguate elements in the chat stream.

For example, the day traders speak in ticker symbols. Providing the content monitor with the URL for the ticker symbol look-up service offered by Yahoo allows the content monitor to weave not only company names but also recent company-related headlines directly into the stream fed to the statistics engine.

While we have focused mainly on chat and bulletin boards, this technology can be applied in other settings. We have begun collaborating with the designers of a natural language interface for Web-based help systems. Here, we give voice to the hundreds of simultaneous conversations taking place between Web site visitors and the automated help system. A similar display can be imagined for other natural language interfaces, including search engines like AskJeeves (www.jeeves.com). In general, the practical applications of this summarization and auditory display tool abound.

3. CONCLUSION AND COMMENTS ON COLLABORATIVE RESEARCH

The two applications outlined in this paper are the first outcomes of a collaboration sponsored by Bell Laboratories and the Brooklyn Academy of Music under the Arts in Multimedia project (AIM). The goal of AIM is to bring together researchers (in this case a statistician) and artists (in this case a sound artist), with the objective of advancing our separate agendas through collaborative projects. Our work together is predicated on the notion that sophistication both in data treatment and aesthetics are crucial to the successful design of audio displays. Thus, in each of our examples, we have endeavored to create a result which communicates information clearly, yet at the same time sounds well composed and appealing. Moving forward, it is our intention to apply these techniques both to practical applications, and also to create a series of artworks. These artworks will use our sonification techniques to establish a series of real-time listening posts, both on the Web and in physical locations. The listening posts will tap in to various points of interest on the Internet, using sound to reveal patterns and trends that would otherwise remain hidden.

In terms of applications, we are exploring the use of sonification to support the design, provisioning and monitoring of communication networks. A network operations center (NOC), for example, routinely receives clues about the health of the system in the form of text messages generated by routers and switches. An audio display installed inside a NOC can act as an early warning system

for approaching bottlenecks as well as aid in troubleshooting. By continued exposure to the sound of a "normally" functioning network, operators will be alerted to system changes that could signal problems.

Art emerges unexpectedly from experimentations with new statistical methods or considerations involving practical applications; and new tools for data analysis and modeling develop in response to artistic concerns. Each of us continues to be surprised by the connections that emerge from rethinking familiar problems in a new context. Through our project, we hope to illustrate both the value of art-technology collaborations as well as their necessity, especially when finding meaning in complex data.

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TAXONOMY AND DEFINITIONS FOR SONIFICATION AND AUDITORY DISPLAY

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ABSTRACT

Sonification is still a relatively young research field and many terms such as sonification, auditory display, auralization, audification have been used without a precise definition. Recent developments such as the introduction of Model-Based Sonification, the establishment of interactive sonification and the increased interest in sonification from arts have raised the need to revisit the definitions in order to move towards a clearer terminology. This paper introduces a new definition for sonification and auditory display that emphasizes the necessary and sufficient conditions for organized sound to be called sonification. It furthermore suggests a taxonomy, and discusses the relation between visualization and sonification. A hierarchy of closed-loop interactions is furthermore introduced. This paper aims to initiate vivid discussion towards the establishment of a deeper theory of sonification and auditory display.

1. INTRODUCTION

Auditory Display is still a young research field whose birth may be perhaps best traced back to the first ICAD conference1 in 1992 organized by Kramer. The resulting proceedings volume “Auditory Display” [1] is still one of the most important books in the field. Since then a vast growth of interest, research, and initiatives in auditory display and sonification has occurred. The potential of sound to support human activity, communication with technical systems and to explore complex data has been acknowledged [2] and the field has been established and has clearly left its infancy.

As in every new scientific field, the initial use of terms lacks coherence and terms are being used with diffuse definitions. As the field matures and new techniques are discovered, old definitions may appear too narrow, or, in light of interdisciplinary applications, too unspecific. This is what motivates the redefinitions in this article.

The shortest accepted definition for sonification is from Barrass and Kramer et al. [2]: “Sonification is the use of non-speech audio to convey information”. This definition excludes speech as this was the primary association in the 1see www.icad.org auditory display of information at that time. The definition is unclear about what is meant by conveyance of information: are real-world interaction sounds sonifications, e.g. of the properties of an object that is being hit? Is a computer necessary for its rendition? As a more specific definition, the definition in [2] continues:

“Sonification is the transformation of data relations into perceived relations in an acoustic signal for the purposes of facilitating communication or interpretation.”

It is significant that the emphasis here is put on the purpose of the usage of sound. This automatically distinguishes sonification from music, where the purpose is not on the precise perception of what interactions are done with an instrument or what data caused the sound, but on an underlying artistic level that operates on a different level. Often, the word ‘mapping’ has been used interchangeably with ‘transformation’ in the above definition. This, however, suggests a severe limitation of sonification towards just mappings between data and sound – which was perfectly fine at the time of the definition where such a ‘Parameter-Mapping Sonification’ was the dominating paradigm.

However, the introduction of Model-Based Sonification (MBS) [3, 4] demonstrates methods to explore data by using sound in a way that is very different from a mapping: in Parameter-Mapping Sonification, data values are mapped to acoustic attributes of a sound (in other words: the data ‘play’ an instrument), whereas in MBS sonification models create and configure dynamic processes that do not make

sound at all without external interactions (in other words: the data is used to build an instrument or sound-capable object, while the playing is left to the user). The user excites the sonification model and receives acoustic responses that are determined by the temporal evolution of the model. By doing this, structural information is holistically encoded into the sound signal, and is no longer a mere mapping of data to sound. One can perhaps state that data are mapped to the configurations of sound-capable objects, but not that they are mapped to sound.

Clearly, sonification models implemented according to MBS are very much in line with the original idea that sonification allows for the discovery of structures in data through sound. Therefore there is the need to reformulate or adapt the definition for sonification to better include such uses of sound, and beyond that hopefully other possible yet-to-be-discovered linkages between data and sound.

Another challenge for the definition comes from the use of sonification in the arts and music: recently more and more artists incorporate methods from sonification in their work. What implications does this have for the term sonification? Think of scientific visualization vs. art: what is the difference between a painting and a modern visualization? Both are certainly organized colors on a surface, both may have aesthetic qualities, yet they operate on a completely different level: the painting is viewed for different layers of interpretation than the visualization. The visualization is expected to have a precise connection to the underlying data, else it would be useless for the process of interpreting the data. In viewing the painting, however, the focus is set more on whether the observer is being touched by it or what interpretation the painter wants to inspire than what can be learnt about the underlying data. Analogies between sonification and music are close-by.

Although music and sonification are both organized sound, and sonifications can sound like music and vice versa, and certainly sonifications can be ‘heard as’ music

as pointed out in [5], there are important differences which are so far not manifest in the definition of sonification.

2. A DEFINITION FOR SONIFICATION

This section introduces a definition for sonification in light of the aforementioned problems. The definition has been refined thanks to many fruitful discussions with colleagues as listed in the acknowledgements and shall be regarded as a new working definition to foster ongoing discussion in the community towards a solid terminology.

Definition: A technique that uses data as input, and generates sound signals (eventually in response to optional additional excitation or triggering) may be called sonification, if and only if

(C1) The sound reflects objective properties or relations in the input data.

(C2) The transformation is systematic. This means that there is a precise definition provided of how the data (and optional interactions) cause the sound to change.

(C3) The sonification is reproducible: given the same data and identical interactions (or triggers) the resulting sound has to be structurally identical.

(C4) The system can intentionally be used with different data, and also be used in repetition with the same data.Data Sonification

Algorithm

systematic

transformation reproducable exchangeability

of data

interactions (optional)

Definition: Sonification

Figure 1: Illustration of the general structure and necessary conditions for sonification. The yellow box depicts besides the sonification elements few other components of auditory displays, see also Sec. 3.

This definition emphasizes important prerequisites for the scientific utility of sonification. It has several partly unexpected implications that are to be explored in the following discussion.

2.1. Discussion

2.1.1. General Comments

Sonification Techniques: According to the above definition, the techniques Audification, Earcons, Auditory Icons, Parameter-Mapping Sonification as well as Model-Based Sonification are all covered by the definition – they all represent information/data by using sound in an organized and well-structured way and they are therefore different sonification technique.2 This may first appear unfamiliar in light of the common parlance to see earcons/auditory icons as different from sonification. However, imagine an auditory display for biomedical data that uses auditory icons as sonic events to represent different classes (e.g. auditory icons for benign/malignant tissue). The sonification would then be the superposition or mixture of all the auditory icons chosen for instance according to the class label and organized properly on the time axis. If we sonify a data set consisting only of a single data item we naturally obtain as an extreme case a single auditory icon. The same can be said for earcons. Although sonification originally has the connotation of representing large and complex data sets, it makes sense for the definition to also work for single data points.

Data vs. Information: A distinction between data and information is – as far as the above definition – irrelevant. Think of earcons to represent computer desktop interactions such as “delete file”, “rename folder”. There can be a lexicon of terms (file, folder, link) and actions (delete, rename, etc.), and in practical computer implementations these features would be represented numerically, e.g. object = O1, action = A3. By doing so, the information has been turned into data, and this is generally done if there is more than one signal type to give. Information like for instance a verbal message can always be represented numerically and thus be understood as data. On the other side, raw data values often carry semantic interpretation: e.g. the outside temperature data value -10◦C (a one-dimensional data set of size 1) – this is cold, and clearly information! Assum-

2they are also covered by the definition of sonification as ‘non-speech use of sound to convey information’!

ing that information is always encoded as data values for its processing we can deal with both in a single definition. How the data are then represented by using sound is another question: whether sonification techniques use a more symbolic or analogic representation according to the analogic-symbolic continuum of Kramer [6] is secondary for the definition.

Mapping as a specific case of sonification: Some articles have used "sonification" to refer specifically to mapping-based sonification, where data features are mapped to acoustic features of sound events or streams. Yet sonification is more generally the representation of data by using sound. There may be times when a clear specification of the sonification technique, e.g. as model-based, audification or parameter-mapping sonification, may be helpful to avoid confusion with the general term of sonification. It makes sense to always use the most specific term possible, that is to use the term Parameter Mapping Sonification, Audification, Model-Based Sonification, etc. to convey exactly what is meant. The term Sonification, however, is, according to the definition, more general which is also supported by many online definitions3. In result we suggest using sonification with the same level of generality as the term visualization is used in visual display.

Sonification as algorithm and sound: Sonification refers to the technique and the process, so basically it refers to the algorithm that is at work between the data, the user and the resulting sound. Often, and with equal right, the resulting sounds are called sonifications. Algorithm means a set of clear rules, independent of whether it is implemented on a computer or any other way.

Sonification as scientific method: According to the definition, sonification is an accurate scientific method which leads to reproducible results, addressing the ear rather then the eye (as visualization does). This does not limit the use of sonifications to data from the sciences, but only states that sonification can be used as a valid instrument to gain insight. The subjectivity in human percep-

3http://en.wikipedia.org/wiki/Sonification, http://wvvel.csee.wvu.edu/sepscor/sonification/lesson9.html,

http://www.techfak.uni-bielefeld.de/ags/ni/projects/datamining/datason/datason e.html, http://www.cs.uiowa.edu/ kearney/22c296Fall02/ CrittendonSpecialty.pdf, to name a few.

tion and interpretation is shared with other perceptualization techniques that bridge the gap between data and the human sensory system. Being a scientific method, a prefix like in "scientific sonification" is not necessary.

Same as some data visualizations may be 'viewed' as art, sonifications may be heard as 'music'[5], yet this use differs from the original intent.

2.1.2. Comments to (C1)

(C1) The sound reflects objective properties or relations in the input data.

Real-world acoustics are typically not a sonification although they often deliver object-property-specific systematic sound, since there is no external input data as requested in C1. For instance, with a bursting bottle, one can identify what is the data, the model and the sound, but the process cannot be repeated with the same bottle. However, using a bottle that fills with rain, hitting it with a spoon once a minute can be seen as a sonification: The data here is the amount of rainfall, which is here measured by the fill level, and the other conditions are also fulfilled. Tuning a guitar string might also be regarded as a sonification to adjust the tension of a string4. These examples show that sonifications are not limited to computer-implementations according to the definition, which embraces the possibility of other non-computer-implemented sonifications.

The borders of sonification and real-world acoustics are fuzzy. It might be discussed how helpful it is to regard or denote everyday sounds as sonifications.

2.1.3. Comments to (C2)

(C2) The transformation is systematic. This means that there is a precise definition provided of how the data (and optional interactions) cause the sound to change.

What exactly do we mean by "precise"? Some sound generators use noise and thereby random elements so that sound events will per se sound different on each rendering. In Parameter-Mapping Sonifications, the intentional addi-

tion of noise (for instance as onset jitter to increase perceptability of events that would otherwise coincide) is often used and makes sense. In order to include such cases randomness is allowed in the definition, yet it is important to declare where and what random elements are used (e.g. by describing the noise distribution). It is also helpful to give a motivation for the use of such random elements. By using too much noise, it is possible to generate useless sonifications in the sense that they garble interpretation of the underlying data. In the same way it is possible to create useless scientific visualizations.

4thanks to the referee for this example!

2.1.4. Comments to (C3)

(C3) The sonification is reproducible: given the same data and identical interactions (or triggers) the resulting sound has to be structurally identical.

The definition claims reproducibility. This may not strictly be achieved for several reasons: the loudspeakers may generate a different sound at different temperatures, other factors such as introduced noise as discussed above may have been added. The use of the term “structurally identical” in the definition aims to weaken the stronger claim of sample-based identity. Sample-based identity is not necessary, yet all possible psychophysical tests should come to identical conclusions.

2.1.5. Comments to (C4)

(C4) The system can intentionally be used with different data, and also be used in repetition with the same data.

Repeatability is essential for a technique to be scientifically valid and useful – otherwise nobody could check the results obtained by using sonification as instrument to gain insight. However, there are some implications by claiming repeatability for what can and cannot be called sonification. It has for instance been suggested that a musician improvising on his instrument produces ‘a sonification of

the musician's emotional state'. With C4, however, "playing a musical instrument" is not a sonification of the performer's emotional state, since it can not be repeated with the 'identical' data. However, the resulting sound may be called a sonification of the interactions with the instrument (regarded here as data), and in fact, music can be heard with the focus to understand the systematic interaction patterns with the instruments.

Some of these conditions have been set as constraints for sonification, e.g. reproducibility in the 'Listening to the Mind Listening' concert5, but not been connected to a definition of sonification.

In summary, the given definition provides a set of necessary conditions for systems and methods to be called sonification. The definition is neither exhaustive nor complete; we hope it will serve as the core definition as we as community work towards a complete one.

3. SONIFICATION AND AUDITORY DISPLAY

With the above definition, the term sonification takes the role of a general term to express the method of rendering

5http://www.icad.org/websiteV2.0/Conferences/ICAD2004/concert call.htm

sound in an organized and well-structured way. This is in good analogy with the term visualization which is also the general term under which a variety of specific techniques such as bar charts, scatter plots, graphs, etc. are subsumed. Particularly there is an analogy between scatter plots where graphical symbols (data-mapped color/size...) are organized in space to deliver the visualization, and Parameter-Mapping Sonification, where in a structurally identical way acoustic events (with data-mapped features) are organized in time. It is helpful to have with sonification a term that operates on the same level of generality as visualization.

This raises the question what then do we mean by auditory displays? Interestingly, in the visual realm, the term 'display' suggests a necessary but complementary part of the interface chain: the device to generate structured light/images, for instance a CRT or LCD display or a projector. So in visualization, the term visualization emphasizes the way how data are rendered as an image while the display is necessary for a user to actually see the information. For

auditory display, we suggest to include this aspect of conversion of sound signals into audible sound, so that an auditory display encompasses also the technical system used to create sound waves, or more general: all possible transmissions which finally lead to audible perceptions for the user. This could range from loudspeakers over headphones to bone conduction devices. We suggest furthermore that auditory display should also include the user context (user, task, background sound, constraints) and the application context, since these are all quite essential for the design and implementation. Sonification is thereby an integral component within an auditory display system which addresses the actual rendering of sound signals which in turn depend on the data and optional interactions, as illustrated in Fig. 2.

Figure 2: Auditory Displays: systems that employ sonification for structuring sound and furthermore include the transmission chain leading to audible perceptions and the application context.

tion since for instance dialogue systems and speech interfaces may also be regarded as auditory displays since they

use sound for communication. While such interfaces are not the primary focus in this research field the terminology suggests their inclusion. On the other hand, Auditory Display may be seen as a subset of the more general term of Auditory Interfaces which do not only include output interfaces (auditory displays, sonification) but also auditory input interfaces which engender bidirectional auditory control and communication between a user and a (in most cases) technical system (e.g. voice control system, query-by humming systems, etc.).

4. HIERARCHY FROM SOUND TO SONIFICATION

So far we have dealt with the necessary conditions surrounding sonification and thus narrowed sonification down to a specific subset of using sound. In this section, we look at sonification in a systemic manner to elucidate its superordinate categories. Figure 3 shows how we suggest to organize the different classes of sound. On the highest level,Map of Sound
Organized Sound
Functional Sounds
Music &
Media Arts Sonification(a)
(b)
Figure 3: Systemic map of sound, showing sonification and its relation to other categories.
sounds are here classified as Organized Sound and unorganized sound. Organized sounds separate from random or otherwise complex structured sounds in the fact that their occurence and structure is shaped by intention. Environmental sounds appear often to be very structured and could thus also be organized sounds, however, if so, any sound would match that category to some extent. It thus may be useful to apply the term to sounds that are intentionally organized – in most cases by the sound/interface developer. The set of organized sound comprises two large sets that partially overlap: music and functional sounds. Music is without question a complex structured signal, organized on various levels, from the acoustic signal to its temporal organization in bars, motifs, parts, layers. It is not our purpose to give a definition of music.

The second set is functional sounds. These are organized sounds that serve a certain function or goal [7]. The function is the motivation for their creation and use. To give an example, all signal sounds (such as telephones, doorbells, horns and warning hooters) are functional sounds. Certainly there are intersections with music, as music can serve functional aspects. For instance, trombones and kettle drums have been used to demonstrate kingship and power. A more subtle function is the use of music in supermarkets to enhance the 'shopping mood'. For that reason these sets overlap – the size of the overlap depends on what is regarded as function.

Sonification in the sense of the above definition is certainly a subset of functional sounds. The sounds are rendered to fulfill a certain function, be it communication of information (signals & alarms), the monitoring of processes, or to support better understanding of structure in data under analysis. So is there a difference between functional sounds and sonification at all? The following example makes clear that sonification is really a subset: Recently a new selective acoustic weapon has been used, the mosquito device6, a loudspeaker that produces a HF-sound inaudible to older people, which drives away teenagers hanging around in front of shops. This sound is surely functional, yet it could neither pass as sonification nor as music.

Finally, we discuss whether sonification has an intersection with music&media arts. Obviously there are many examples where data are used to drive aspects of musical performances, e.g. data collected from motion tracking or biosensors attached to a performer. This is, concerning the involved techniques and implementations similar to mapping sonifications. However, a closer look at our proposed definition shows that often the condition for the transformation to be systematic C2 is violated and the exact rules are not made explicit. But without making the relationship explicit, the listener cannot use the sound to understand the underlying data better. In addition, condition C4 may often be violated. If sonification-like techniques are employed to obtain a specific musical or acoustic effect without transparency between the used data and details of the sonification tech-

niques, it might, for the sake of clarity, better be denoted as ‘data-inspired music’, or ‘data-controlled music’ than as sonification. Iannis Xenakis, for instance, did not even want the listener to be aware of the data source nor the rules of sound generation.

6see http://www.compoundsecurity.co.uk/, last seen 2008-01-16

5. CLOSED INTERACTION LOOPS IN AUDITORY DISPLAYS

This section emphasizes the role of interaction in sonification. We propose different terms depending on the scope of the closure of the interaction loop. The motivation for this discussion is that it might be helpful to address how terms such as biofeedback or interactive sonification relate to each other.

We start the discussion with Fig. 4 that depicts closed loop interactions. The sonification module in the upper center playing rendered sonifications to the user. Data sources for sonification enter the box on the left side and the most important parts are (a) World/System: this comprises any system in the world that is connected to the sonification module, e.g. via sensors that measure its state, and (b) Data: these are any data under analysis or represented information to be displayed that are stored separately and accessible by the sonification.World/System
Sonification
Interactive Sonification
Human Activity (supported by sonification)
Auditory Biofeedback
Data
Navigation
Monitoring
No Action

Figure 4: Illustration of Closed-Loop Auditory Systems.

In this setting, Process Monitoring is the least interactive sonification, where data recorded from the world (in real-time) or read from the data repository is continuously used as input for a sonification rendering process. Here, the

listener is merely passively listening to the sound with the only active component being his/her focus of attention onto parts of the sound. Certainly, certain changes in the sound might attract attention and force the user to act (e.g. sell stocks, stop a machine, etc...).

A higher degree of active involvement occurs when the user actively changes and adjusts parameters of the sonification module, or interacts otherwise with the sonification system. We denote this case as Interactive Sonification. There is a wide field of possibilities of why and how to do so, and we discuss 3 different prototypical examples:

(a) Triggering: Consider a mapping sonification of a given data set. An essential interaction for the user is to issue the command to render/playback the sonification for a selected dataset. Possibly he/she does this several times in order to attend different parts of the sound signal. This elementary case is an interaction, however, a very basic one.

(b) Parameter Adjustment is done when the user changes parameters, such as what data feature are mapped to acoustic parameters, control ranges, compression factors, etc. Often such adjustments happen separate from the playback so that the changes are made and afterwards the updated sound is rendered. However, interactive real-time control is feasible in many cases and shows a higher degree of interactivity. The user actively explores the data by generating different 'views' of the data [8]. In visualization a similar interactivity is obtained by allowing the user to select axes scalings, etc.

(c) Excitatory Interaction is the third sort of interaction and is structurally similar to the case of triggering. Particularly in Model-Based Sonification [4], usually the data are used to configure a sound-capable virtual object that in turn reacts on excitatory interactions with acoustic responses whereby the user can explore the data interactively. Excitation puts energy into the dynamic system and thus initiates an audible dynamical system behavior. Beyond a simple triggering, excitatory interactions can be designed to make

use of the fine-grained manipulation skills that human hands allow, e.g. by enabling to shake, squeeze, tilt or deform the virtual object, for instance using sensor-equipped physical interfaces to interact with the sonification model. A good example for MBS is Shoogle by Williamson et al. [9], where short text messages in a mobile phone can be overviewed by shaking a mobile phone equipped with accelerometer sensors, resulting in audible responses of the text messages as objects moving virtually inside the phone. Excitatory interactions offer rich and complex interactions for interactive sonification.

The next possibility for a closed loop is by interactions that select or browse data. Since data are chosen, it may best be referred to as Navigation. Navigation can also be regarded as special case of Interactive Sonification, depending on where the data are selected and the borders are here really soft. Navigation usually goes hand in hand with triggering of sonification (explained above).

Auditory Biofeedback can be interpreted as a sonification of measured sensor data. In contrast to the above types, the user's activity is not controlling an otherwise autonomous sonification with independent data, but it produces the input data for the sonification system. The user perceives a sound that depends on his/her own activity. Such systems have applications that range from rehabilitation training to movement training in sports, e.g. to perform

a complex motion sequence (e.g. a tennis serve) so that its sonification is structurally more similar to the sonification of an expert performing the action [10].

The final category is Human Activity, which means that the interaction ranges beyond the sonification system into the world, often driven by the goal to change a world state in a specific way. In turn, any sensors that pick up the change may lead to changes in the sonification. The difference between the loop types before is that the primary focus is to achieve a goal beyond the sonification system, and

not to interact with a closed-loop sonification system. Even without attending the sonification consciously or primarily, the sound can be helpful to reach the goal. For example, imagine the real-world task to fill a thermos bottle with tea. While your primary goal is to get the bottle filled you will receive the 'gluck-gluck' sound with increasing pitch as a by-product of the interaction. If this is consistently useful, you subconsciously adapt your activity to exploit the cues in the sound – but the sound is only periphery for the goal. In a similar sense, sonifications may deliver helpful by-products to actions that change the world state. We regard such interaction add-ons where sonification is a non-obtrusive yet helpful cue for goal attainment as inspiring design direction. Such sonifications might even become subliminal in the sense that users, when asked about the sound, are not even aware of the sound, yet they perform better with sound than without.

6. DISCUSSION AND CONCLUSION

The definitions in this paper are given on the basis of three goals: (i) to anchor sonification as a precise scientific method so that it delivers reproducible results and thus can be used and trusted as instrument to obtain insight into data under analysis. (ii) to offer a generalization which does not limit itself to the special case of mappings from data to sound, but which introduces sonification as general systematic mediator between data and sound, whatever the representation might be. (iii) to balance the definition so that the often-seen pair of terms 'visualization & sonification' are at the same level of generality.

The definition has several implications which have been discussed in Sec. 2. We'd like to emphasize that this effort is being done in hope that the definition inspires a general discussion on the terminology and taxonomy of the research field of auditory display. An online version of the definition is provided at www.sonification.de with the aim to collect comments and examples of sonifications as well as examples that are agreed not to be sonifications and which help in turn to improve the definition.

In Section 3, we described integral parts for auditory display so that sonification takes a key component as the

technical part involving the rendition of sound. Again, the suggested modules are meant as working hypothesis to be discussed at ICAD.

While the given definitions specified terms on a horizontal level, Section 4 proposes a vertical organization of sound in relation to often used terms. The intersections between the different terms and categories have been addressed with examples.

Finally, we have presented in Section 5 an integrative scheme for organizing different classes of auditory closed loops according to the loop closure scope. It proves helpful to clarify classes of interactive sonifications. We think that grouping existing sonifications according to these categories can be helpful to better find alternative approaches for a given task.

The suggested terminology and taxonomy is the result of many discussions and a thorough search for helpful concepts. We suggest it as working definitions to be discussed at the interdisciplinary level of ICAD in hope to contribute towards a maturing of the fields of auditory display and sonification.

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Nomadic Radio: Scaleable and Contextual Notification for Wearable Audio Messaging.
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Nomadic Radio: Scaleable and Contextual Notification for Wearable Audio Messaging
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ABSTRACT
Mobile workers need seamless access to communication

and information services on portable devices. However current solutions overwhelm users with intrusive and ambiguous notifications. In this paper, we describe scaleable auditory techniques and a contextual notification model for providing timely information, while minimizing interruptions. User's actions influence local adaptation in the model. These techniques are demonstrated in Nomadic Radio, an audio-only wearable computing platform.
Keywords
Auditory I/O, passive awareness, wearable computing, adaptive interfaces, interruptions, notifications
INTRODUCTION
In today's information-rich environments, people use a number of appliances and portable devices for a variety of tasks in the home, workplace and on the run. Such devices are ubiquitous and each plays a unique functional role in a user's lifestyle. To be effective, these devices need to notify users of changes in their functional state, incoming messages or exceptional conditions. In a typical office environment, the user attends to a plethora of devices with notifications such as calls on telephones, asynchronous messages on pagers, email notification on desktop computers, and reminders on personal organizers or watches. This scenario poses a number of key problems.
Lack of Differentiation in Notification Cues
Every device provides some unique form of notification. In many cases, these are distinct auditory cues. Yet, most cues are generally binary in nature, i.e. they convey only the occurrence of a notification and not its urgency or dynamic state. This prevents users from making timely decisions about received messages without having to shift focus of attention (from the primary task) to interact with the device and access the relevant information.

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Minimal Awareness of the User and Environment

Such notifications occur without any regard to the user's engagement in her current activity or her focus of attention. This interrupts a conversation or causes an annoying disruption in the user's task and flow of thoughts. To prevent undue embarrassment in social environments, users typically turn off cell-phones and pagers in meetings or lectures. This prevents the user from getting notification of timely messages and frustrates people trying to get in touch with her.

No Learning from Prior Interactions with User

Such systems typically have no mechanism to adapt their behavior based on the positive or negative actions of the user. Pagers continue to buzz and cell-phones do not stop ringing despite the fact that the user may be in a conversation and ignoring the device for some time.

Lack of Coordinated Notifications

All devices compete for a user's undivided attention without any coordination and synchronization of their notifications. If two or more notifications occur within a short time of each other, the user gets confused or frustrated. As people start carrying around many such portable devices, frequent and uncoordinated interruptions inhibit their daily tasks and interactions in social environments.

Given these problems, most devices fail to serve their intended purpose of notification or communication, and thus do not operate in an efficient manner for a majority of their life cycle. New users choose not to adopt such technologies, having observed the obvious problems encountered with their usage. In addition, current users tend to turn off the devices in many situations, inhibiting the optimal operation of such personal devices.

Nature of Interruptions in the Workplace

A recent observational study [4] evaluated the effect of interruptions on the activity of mobile professionals in their workplace. An interruption, defined as an asynchronous and unscheduled interaction, not initiated by the user, results in the recipient discontinuing the current activity. The results revealed several key issues. On average, sub.jects were

interrupted over 4 times per hour, for an average duration slightly over 2 minutes. Hence, nearly 10 minutes per hour

was spent on interruptions. Although a majority of the interruptions occurred in a face-to-face setting, 20% were due to telephone calls (no email or pager activity was analyzed in this study). In 64% of the interruptions, the recipient received some benefit from the interaction. This suggests that a blanket approach to prevent interruptions, such as holding all calls at certain times of the day, would prevent beneficial interactions from occurring. However in 41% of the interruptions, the recipients did not resume the work they were doing prior to it. But active use of new communication technologies makes users easily vulnerable to undesirable interruptions.

These interruptions constitute a significant problem for mobile professionals using tools such as pagers, cell-phones and PDAs, by disrupting their time-critical activities. Improved synchronous access using these tools benefits initiators but leaves recipients with little control over the interactions. The study suggests development of improved filtering techniques that are especially light-weight, i.e. don't require more attention from the user and are less disruptive than the interruption itself. By moving interruptions to asynchronous media, messages can be stored for retrieval and delivery at more appropriate times.

NOMADIC RADIO: WEARABLE AUDIO MESSAGING

Personal messaging and communication, demonstrated in Nomadic Radio, provides a simple and constrained problem domain in which to develop and evaluate a contextual notification model. Messaging requires development of a model that dynamically selects a suitable notification strategy based on message priority, usage level, and environmental context. Such a system must infer the user's attention by monitoring her current activities such as interactions with the device and conversations in the room. The user's prior responses to notifications must also be taken into consideration to adapt the notifications over time. In this paper, we will consider techniques for scaleable

auditory presentation and an appropriate parameterized approach towards contextual notification.

Several recent projects utilized speech and audio I/O on wearable devices to present information. A prototype augmented audio tour guide [l] played digital audio recordings indexed by the spatial location of visitors in a museum. SpeechWear [11] enabled users to perform data entry and retrieval using speech recognition and synthesis. Audio Aura [10] explored the use of background auditory cues to provide serendipitous information coupled with people's physical location in the workplace. In Nomadic Radio, the user's inferred context rather than actual location is used to decide when and how to deliver scaleable audio notifications. In a recent paper [13], researchers suggest the use of sensors and user modeling to allow wearables to infer when users should be interrupted by incoming messages. They suggest waiting for a break in the conversation to post a message summary on the user's heads-up display. In this paper we describe a primarily non-visual approach to provide timely information to nomadic listeners, based on a variety of contextual cues.

Nomadic Radio is a wearable computing platform that provides a unified audio-only interface to remote services and messages such as email, voice mail, hourly news broadcasts, and personal calendar events. These messages are automatically downloaded to the device throughout the day and users can browse through them using voice commands and tactile input. The system consists of Java-based clients and remote servers (written in C and Perl) that communicate over wireless LAN, and utilize the telephony infrastructure in the Speech Interface group. Simultaneous spatial audio streams are rendered using a HRTF-based Java audio API. Speech I/O is provided via a networked implementation of AT&T Watson Speech API.

To provide a hands-free and unobtrusive interface to a nomadic user, the system primarily operates as a wearable audio-only device. The SoundBeam Neckset, a research prototype patented by Nortel for use in hands-free telephony, was adapted as the primary wearable platform in Nomadic Radio. It consists of two directional speakers

mounted on the user's shoulders, and a directional microphone placed on the chest (see figure 1). Here information and feedback is provided to the user through a combination of auditory cues, spatial audio rendering, and synthetic speech. Integration of a variety of auditory techniques on a wearable device provides hands-free access and navigation as well as lightweight and expressive notification.

An audio-only interface has been incorporated in Nomadic Radio, and a networked infrastructure for unified messaging has been developed for wearable access [12]. The system currently operates on a Libretto 100 mini-portable PC worn by the user. The key issue addressed in this paper is that of handling interruptions to the listener in a manner that reduces disruption, while providing timely notifications for contextually relevant messages.

USAGE AND NOTIFICATION SCENARIO

The following scenario demonstrates the audio interface and presentation of notifications in Nomadic Radio (no voice commands from the user are shown here).

SCALEABLE AUDITORY PRESENTATION

A scaleable presentation is necessary for delivering sufficient information while minimizing interruption to the listener. Messages in Nomadic Radio are scaled dynamically to unfold as seven increasing levels of notification (see figure 3): silence, ambient cues, auditory cues, message summary, preview, full body, and foreground rendering. These are described further below:

Silence for Least Interruption and Conservation

In this mode all auditory cues and speech feedback are turned-off. Messages can be scaled down to silence when the message priority is inferred to be too low for the message to be relevant for playback or awareness to a user, based on her recent usage of the device and the conversation level. This mode also serves to conserve processing, power and memory resources on a portable device or wearable computer.

Ambient Cues for Peripheral Awareness

In Nomadic Radio, ambient auditory cues are continuously played in the background to provide an awareness of the operational state of the system and ongoing status of messages being downloaded (see figure 4). The sound of flowing water provides an unobtrusive form of ambient awareness that indicates the system is active (silence indicates sleep mode). Such a sound tends to fade into the perceptual background after a short time, so it does not distract the listener. The pitch is increased during file downloads, momentarily foregrounding the ambient sound. A short e-mail message sounds like a splash while a two-minute audio news summary is heard as faster flowing water while being downloaded. This implicitly indicates message size without the need for additional audio cues and prepares the listener to hear (or deactivate) the message before it becomes available. Such peripheral awareness minimizes cognitive overhead of monitoring incoming messages relative to notifications played as distinct auditory cues, which incur a somewhat higher cost of attention on part of the listener.

Related Work in Auditory Awareness

In ARKola [5], an audio/visual simulation of a bottling factory, repetitive streams of sounds allowed people to keep track of activity, rate, and functioning of running machines. Without sounds people often overlooked problems; with auditory cues, problems were indicated by the machine's sound ceasing (often ineffective) or via distinct alert sounds. The various auditory cues (as many as 12 sounds play simultaneously) merged as an auditory texture, allowed people to hear the plant as a complex integrated process. Background sounds were also explored in ShareMon [3], a prototype application that notified users of file sharing activity. Cohen found that pink noise used to indicate %CPU time was considered "obnoxious", even though users understood the, pitch correlation. However, preliminary reactions to wave sounds were considered positive and even soothing. In Audio Aura [IO], alarm

sounds were eliminated and a number of "harmonically coherent sonic ecologies" were explored, mapping events to

auditory, musical or voice-based feedback. Such techniques were used to passively convey the number of email messages received, identity of senders, and abstract representations of group activity.

Auditory Cues for Notification and Identification

In Nomadic Radio, auditory cues are a crucial means for conveying awareness, notification and providing necessary assurances in its non-visual interface. Different types of auditory techniques provide distinct feedback, awareness and message information.

Feedback Cues

Several types of audio cues indicate feedback for a number of operational events in Nomadic Radio:

1. Task completion and confirmations - button pressed, speech understood, connected to servers, finished playing or loaded/deleted messages.
2. Mode transitions - switching categories, going to non-speech or ambient mode.
3. Exceptional conditions - message not found, lost connection with servers, and errors.

Priority Cues for Notification

In a related project, “email glances” [7] were formulated as a stream of short sounds indicating category, sender and content flags (from keywords in the message). In Nomadic Radio, message priority inferred from email content filtering provides distinct auditory cues (assigned by the user) for group, personal, timely, and important messages. In addition, auditory cues such as telephone ringing indicate voice mail, whereas an extracted sound of a station identifier indicates a news summary.

VoiceCues for Identification

VoiceCues represent a novel approach for easy identification of the sender of an email, based on a unique auditory signature of the person. VoiceCues are created by manually extracting a l-2 second audio sample from the voice messages of callers and associating them with their respective email login. When a new email message arrives, the system queries its database for a related VoiceCue for that person before playing it to the user as a notification, along with the priority cues. The authors have found

VoiceCues to be a remarkably effective method for quickly conveying the sender of the message in a very short duration. This technique reduces the need for synthetic speech feedback, which can often be distracting.

Message Summary Generation

A spoken description of an incoming message can present relevant information in a concise manner. Such a description typically utilizes header information in email messages to convey the name of the sender and the subject of the message. In Nomadic Radio, message summaries are generated for all messages, including voice-mail, news and calendar events. The summaries are augmented by additional attributes of the message indicating category, order, priority, and duration. For audio sources, like voice messages and news broadcasts, the system plays the first 2.5 seconds of the audio. This identifies the caller and the urgency of the call, inferred from intonation in the caller's voice or provides a station identifier for news summaries.

Message Previews using Content Summarization

Messages are scaled to allow listeners to quickly preview the contents of an email or voice message. In Nomadic Radio, a preview for text messages extracts the first 100 characters of the message (a default size that can be user defined). This heuristic generally provides sufficient context for the listener to anticipate the overall message theme and urgency. For email messages, redundant headers and previous replies are eliminated from the preview for effective extraction. Use of text summarization techniques, based on tools such as ProSum' developed by British Telecom, would allow more flexible means of scaling message content. Natural language parsing techniques used in ProSum permit a scaleable summary of an arbitrarily large text document.

A preview for an audio source such as a voice message or news broadcast presents a fifth of the message at a gradually increasing playback rate of up to 1.3 times faster than normal. There are a range of techniques for time-compressing speech without modifying the pitch, however

twice the playback rate usually makes the audio incomprehensible. A better representation for content summarization requires a structural description of the audio, based on annotated or automatically determined pauses in speech, speaker and topic changes. Such an auditory thumbnail must function similar to its visual counterpart. A preview for a structured voice message would provide pertinent aspects such as name of caller and phone number, whereas a structured news preview would be heard as the hourly headlines.

Full Body: Playing Complete Message Content

This mode plays the entire audio file or reads the full text of the message at the original playback rate. Some parsing of the text is necessary to eliminate redundant header information and format tags. The message is augmented with summary information indicating sender and subject. This message is generally spoken or played in the background of the listener's audio space.

I http://transend.Iabs.bt.com/prosum/on-line/

Foreground Rendering via Spatial Proximity

An important message is played in the foreground of the listening space. The audio source of the message is rapidly moved closer to the listener, allowing it to be heard louder, and played there for 415" of its duration. The message gradually begins to fade away, moving back to its original position and amplitude for the remaining l/S" of the duration. The foregrounding algorithm ensures that the messages are quickly brought into perceptual focus by pulling them to the listener rapidly. However the messages are pushed back slowly to provide an easy fading effect as the next one is heard. As the message moves its spatial direction is maintained so that the listener can retain a focus on the audio source even if another begins to play.

Hence a range of techniques provide scaleable forms of background awareness, auditory notification, spoken feedback and foreground rendering of incoming messages.

CONTEXTUAL NOTIFICATION

In Nomadic Radio, context dynamically scales the notifications for incoming messages. The primary contextual cues used include: message priority from email

filtering, usage level based on time since last user action, and the likelihood of conversation estimated from real-time analysis of the auditory scene. In our experience these parameters provide sufficient context to scale notifications, however data from motion or location sensors can also be integrated in such a model. A linear and scaleable auditory notification model is utilized, based on the notion of estimating costs of interruption and the value of information to be delivered to the user. This approach is similar to recent work [6] on using perceptual costs and a focus of attention model for scaleable graphics rendering.

Message Priority

The priority of incoming messages is explicitly determined via content-based email filtering using CLUES [9], a filtering and prioritization system. CLUES has been integrated into Nomadic Radio to determine the timely nature of messages by finding correlation between a user's calendar, rolodex, to-do list, as well as a record of outgoing messages and phone calls. These rules are integrated with static rules created by the user for prioritizing specific people or message subjects. When a new email message arrives, keywords from its sender and. subject header information are correlated with static and generated filtering rules to assign a priority to the message. Email messages are also prioritized if the user is traveling and meeting others in the same geographic area (via area codes in the rolodex). The current priorities include: group, personal, very important, most important, and timely. Priorities are parameterized by logarithmically scaling all priorities within a range of 0 to 1. Logarithmic scaling ensures that higher priority messages are weighted higher relative to unimportant or uncategorized messages.

Priority (i) = (log (i) / log (Priority Levels Mu))

Usage Level

One problem with using last actions for setting usage levels is that if a user deactivates an annoying message, that action is again time-stamped. Such negative reinforcements continue to increase the usage level and the related

notification. Therefore negative actions such as stopping audio playback or deactivating speech are excluded from generating actions for computing the usage.

Likelihood of Conversation

Conversation in the environment can be used to gauge whether the user is in a social context where an interruption is less appropriate. If the system detects the occurrence of more than several speakers over a period of time, that is an indication of a conversational situation. Auditory events are first detected by adaptively thresholding total energy and incorporating constraints on event length and surrounding pauses. The system uses mel-scaled filter-bank coefficients (MFCs) and pitch estimates to discriminate, reasonably well, a variety of speech and non-speech sounds. HMMs (Hidden Markov Models) capture both the temporal characteristics and spectral content of sound events. The techniques for feature extraction and classification of the auditory scene using HMMs are described in a recent workshop paper [2]. The likelihood of speech detected in the environment is computed for each event in a short window of time. In addition, the probabilities are weighted, such that most recent time periods in the window are considered more relevant for computing the overall Speech Level. We are evaluating the classifier's effectiveness by training it with a variety of speakers and background sounds.

Notification Level

A weighted average for all three contextual cues provides level has an inversely proportional relationship with notification i.e. a lower notification must be provided during high conversation.

Presentation Latency

Latency represents the period of time to wait before playing the message to the listener, after a notification cue is delivered. Latency is computed as a function of the notification level and the maximum window of time (Latency,& that a lowest priority message can be delayed for playback. The default maximum latency is set to 20 seconds, but can be modified by the user.

were increased. Jane was notified of a group message shortly after the voice message, since the system detected higher usage activity. Hence, the system correctly scaled down notifications when Jane did not want to be bothered whereas notifications were scaled up when Jane started to use the system to browse her messages.

EFFECTIVENESS OF THE NOTIFICATION MODEL

The nature of peripheral awareness and unobtrusive notification on a wearable device requires a usage evaluation that must be conducted on an ongoing and long-term basis. However, the predictive effectiveness of the notification model must first be evaluated on a quantitative basis. Hence, all message and notification parameters are captured for such analysis. Lets consider two actual examples of notification computed for email messages with different priorities. Figure 7 shows an auditory cue generated for a group message (low priority).

The timely message (in figure 8) received greater priority and consequently a higher notification level for summary playback. A moderate latency time (approx. 6 secs.) was chosen. However when the user interrupted the notification by a button press, the summary playback was aborted. The user's action reduced overall weights by 5%.

Dynamic Adaptation of the Notification Model

The user can initially set the weights for the notification model to high, medium, or low (interruption). These weight settings were selected by experimenting with notifications over time using an interactive visualization of message parameters. This allowed us to observe the model, modify weights and infer the effect on notification based on different weighting strategies. Pre-defined weights provide an approximate behavior for the model and help bootstrap the system for novice users. The system also allows the user to dynamically adjust these weights (changing the interruption and notification levels) by their implicit actions while playing or ignoring messages.

The system allows localized positive and negative reinforcement of the weights by monitoring the actions of

the user during notifications. As a message arrives, the system plays an auditory cue if its computed notification level is above the necessary threshold for auditory cues. It then uses the computed latency interval to wait before playing the appropriate summary or preview of the message. During that time, the user can request the message be played earlier or abort any further notification for the message via speech or button commands. If aborted, all weights are reduced by a fixed percentage (default is 5%), a negative reinforcement. If the user activates the message (positive reinforcement) within 60 seconds after the notification, the playback scale selected by the user is used to increase all weights. If the message is ignored, no change is made to the weights, but the message remains active for 60 seconds during which the user's actions can continue to influence the weights.

Figure 6 shows a zoomed view of the extended scenario introduced earlier, focusing on Jane's actions that reinforce the model. Jane received several messages and ignored most of the group messages and a recent personal message (the weights remain unchanged). While in the meeting, Jane interrupted a timely message to abort its playback. This reduced the weights for future messages, and the ones with low priority (group message) were not notified to Jane. The voice message from Kathy, her daughter, prompted Jane to reinforce the message by playing it. In this case, the weights

Continuous local reinforcement over time should allow the system to reach a state where it is somewhat stable and robust in converging to the user's preferred notification. Currently the user's actions primarily adjust weights for subsequent messages, however effective reinforcement learning requires a model that generalizes a notification policy that maximizes some long-term measure of reinforcement [8]; this will be the focus of our future work.

PRELIMINARY EVALUATION

Although the authors have been using and relining these techniques during system development, a preliminary 2-day evaluation was conducted with a novice user, who had prior experience with mobile phones and 2-way pagers. The user

was able to listen to notifications while attending to tasks in parallel such as reading or typing. He managed to have casual discussions with others while hearing notifications; however he preferred turning off all audio during an important meeting with his advisor. People nearby sometimes found the spoken feedback distracting if heard louder, however that also cued them to wait before interrupting the user. The volume on the device was lowered to minimize any disruption to others and maintain the privacy of messages. The user requested an automatic volume gain that adapted to the environmental noise level. In contrast to speech-only feedback, the user found the unfolding presentation of ambient and auditory cues allowed sufficient time to switch attention to the incoming message. Familiarization with the auditory cues was necessary. He preferred longer and gradual notifications rather than distinct auditory tones. The priority cues were the least useful indicator whereas VoiceCues provided obvious benefit. Knowing the actual priority of a message was less important than simply having it presented in the right manner. The user suggested weaving message priority into the ambient audio (as increased pitch). He found the overall auditory scheme somewhat complex, preferring instead a simple notification consisting of ambient awareness, Voice&es and spoken text.

The user stressed that the ambient audio provided the most benefit while requiring least cognitive effort. He wished to hear ambient audio at all times to remain reassured that the system was still operational. An unintended effect discovered was that a “pulsating” audio stream indicated low battery power on the wearable device. A “pause” button was requested, to hold all messages while participating in a conversation, along with subtle but periodic auditory alerts for unread messages waiting in queue. The user felt that Nomadic Radio provided appropriate awareness and its expressive qualities justified its use over a pager. A long-term trial with several nomadic users is necessary to further validate these notification techniques.

CONCLUSIONS

We have demonstrated techniques for scaleable auditory

presentation and message notification using a variety of contextual cues. The auditory techniques and notification model have been refined based on continuous usage by the authors, however we are currently conducting additional evaluations with several users. Ongoing work explores adaptation of the notification model based on reinforcement from user behavior over time. Our efforts have focused on wearable audio platforms, however these ideas can be readily utilized in consumer devices such as pagers, PDAs and mobile phones to minimize disruptions while providing timely information to users on the move.

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The Physics of Sound

Sound lies at the very center of speech communication. A sound wave is both the end product of the speech production mechanism and the primary source of raw material used by the listener to recover the speaker's message.

Because of the central role played by sound in speech communication, it is important to have a good understanding of how sound is produced, modified, and measured. The purpose of this chapter will be to review some basic principles underlying the physics of sound, with a particular focus on two ideas that play an especially important

role in both speech and hearing: the concept of the spectrum and acoustic filtering. The speech production
mechanism is a kind of assembly line that operates by generating some relatively simple sounds consisting of
various combinations of buzzes, hisses, and pops, and then filtering those sounds by making a number of fine
adjustments to the tongue, lips, jaw, soft palate, and other articulators. We will also see that a crucial step at the
receiving end occurs when the ear breaks this complex sound into its individual frequency components in much the
same way that a prism breaks white light into components of different optical frequencies. Before getting into these
ideas it is first necessary to cover the basic principles of vibration and sound propagation.

Sound and Vibration

A sound wave is an air pressure disturbance that results from vibration. The vibration can come from a tuning
fork, a guitar string, the column of air in an organ pipe, the head (or rim) of a snare drum, steam escaping from a
radiator, the reed on a clarinet, the diaphragm of a loudspeaker, the vocal cords, or virtually anything that vibrates in
a frequency range that is audible to a listener (roughly 20 to 20,000 cycles per second for humans). The two
conditions that are required for the generation of a sound wave are a vibratory disturbance and an elastic medium,
the most familiar of which is air. We will begin by describing the characteristics of vibrating objects, and then see
what happens when vibratory motion occurs in an elastic medium such as air. We can begin by examining a simple
vibrating object such as the one shown in Figure 3-1. If we set this object into vibration by tapping it from the
bottom, the bar will begin an upward and downward oscillation until the internal resistance of the bar causes the
vibration to cease.

The graph to the right of Figure 3-1 is a visual representation of the upward and downward motion of the bar.

To see how this graph is created, imagine that we use a strobe light to take a series of snapshots of the bar as it
vibrates up and down. For each snapshot, we measure the instantaneous displacement of the bar, which is the

difference between the position of the bar at the split second that the snapshot is taken and the position of the bar at
rest. The rest position of the bar is arbitrarily given a displacement of zero; positive numbers are used for
displacements above the rest position, and negative numbers are used for displacements below the rest position. So,
the first snapshot, taken just as the bar is struck, will show an instantaneous displacement of zero; the next snapshot
will show a small positive displacement, the next will show a somewhat larger positive displacement, and so on. The
pattern that is traced out has a very specific shape to it. The type of vibratory motion that is produced by a simple
vibratory system of this kind is called simple harmonic motion or uniform circular motion, and the pattern that is
traced out in the graph is called a sine wave or a sinusoid.

Figure 3-1. A bar is fixed at one and is set into vibration by tapping it from the bottom. Imagine that
a strobe light is used to take a series of snapshots of the bar as it vibrates up and down. At each
snapshot the instantaneous displacement of the bar is measured. Instantaneous displacement is the
distance between the rest position of the bar (defined as zero displacement) and its position at any
particular instant in time. Positive numbers signify displacements that are above the rest position,
while negative numbers signify displacements that are below the rest position. The vibratory pattern
that is traced out when the sequence of displacements is graphed is called a sinusoid.

Basic Terminology

We are now in a position to define some of the basic terminology that applies to sinusoidal vibration.

periodic: The vibratory pattern in Figure 3-1, and the waveform that is shown in the graph, are examples of
periodic vibration, which simply means that there is a pattern that repeats itself over time.

cycle: Cycle refers to one repetition of the pattern. The instantaneous displacement waveform in Figure 3-1 shows
four cycles, or four repetitions of the pattern.

period: Period is the time required to complete one cycle of vibration. For example, if 20 cycles are completed in 1
second, the period is 1/20th of a second (s), or 0.05 s. For speech applications, the most commonly used unit of

measurement for period is the millisecond (ms):
1 ms = 1/1,000 s = 0.001 s = 10 -3 s
A somewhat less commonly used unit is the microsecond (μs):
1 μs = 1/1,000,000 s = 0.000001 s = 10 -6 s
frequency: Frequency is defined as the number of cycles completed in one second. The unit of measurement for
frequency is hertz (Hz), and it is fully synonymous the older and more straightforward term cycles per second
(cps). Conceptually, frequency is simply the rate of vibration. The most crucial function of the auditory system is to
serve as a frequency analyzer – a system that determines how much energy is present at different signal frequencies.
Consequently, frequency is the single most important concept in hearing science. The formula for frequency is:
f = 1/t, where: f = frequency in Hz
t = period in seconds
So, for a period 0.05 s:
f = 1/t = 1/0.05 = 20 Hz
It is important to note that period must be represented in seconds in order to get the answer to come out in cycles per
second, or Hz. If the period is represented in milliseconds, which is very often the case, the period first has to be
converted from milliseconds into seconds by shifting the decimal point three places to the left. For example, for a
period of 10 ms:
f = 1/10 ms = 1/0.01 s = 100 Hz
Similarly, for a period of 100 μs:
f = 1/100 μs = 1/0.0001 s = 10,000 Hz
The period can also be calculated if the frequency is known. Since period and frequency are inversely related, t
= 1/f. So, for a 200 Hz frequency, t = 1/200 = 0.005 s = 5 ms.
Characteristics of Simple Vibratory Systems
Simple vibratory systems of this kind can differ from one another in just three dimensions: frequency,
amplitude, and phase. Figure 3-2 shows examples of signals that differ in frequency. The term amplitude is a bit
different from the other terms that have been discussed thus far, such as force and pressure. As we saw in the last
chapter, terms such as force and pressure have quite specific definitions as various combinations of the basic

dimensions of mass, time, and distance. Amplitude, on the other hand, will be used in this text as a generic term
meaning "how much." How much what? The term amplitude can be used to refer to the magnitude of displacement,
the magnitude of an air pressure disturbance, the magnitude of a force, the magnitude of power, and so on. In the

present context, the term amplitude refers to the magnitude of the displacement pattern. Figure 3-3 shows two
displacement waveforms that differ in amplitude. Although the concept of amplitude is as straightforward as the two
waveforms shown in the figure suggest, measuring amplitude is not as simple as it might seem. The reason is that
the instantaneous amplitude of the waveform (in this case, the displacement of the object at a particular split
second in time) is constantly changing. There are many ways to measure amplitude, but a very simple method called
peak-to-peak amplitude will serve our purposes well enough. Peak-to-peak amplitude is simply the difference in
amplitude between the maximum positive and maximum negative peaks in the signal. For example, the bottom
panel in Figure 3-3 has a peak-to-peak amplitude of 10 cm, and the top panel has a peak-to-peak amplitude of 20
cm. Figure 3-4 shows several signals that are identical in frequency and amplitude, but differ from one another in

phase. The waveform labeled 0 o phase would be produced if the bar were set into vibration by tapping it from the
bottom. The waveform labeled 180 o phase would be produced if the bar were set into vibration by tapping it from
the top, so that the initial movement of the bar was downward rather than upward. The waveforms labeled 90 o phase
and 270 o phase would be produced if the bar were set into vibration by pulling the bar to maximum displacement
and letting go -- beginning at maximum positive displacement for 90 o phase, and beginning at maximum negative
displacement for 270 o phase. So, the various vibratory patterns shown in Figure 3-4 are identical except with respect
to phase; that is, they begin at different points in the vibratory cycle. As can be seen in Figure 3-5, the system for
representing phase in degrees treats one cycle of the waveform as a circle; that is, one cycle equals 360 o. For
example, a waveform that begins at zero displacement and shows its initial movement upward has a phase of 0 o, a
waveform that begins at maximum positive displacement and shows its initial movement downward has a phase of
90 o, and so on.

Figure 3-2. Two vibratory patterns that differ in frequency. The panel on top is higher in frequency
than the panel on bottom.

Figure 3-3. Two vibratory patterns that differ in amplitude. The panel on top is higher in amplitude than the
panel on bottom.
Phase: 0
Phase: 90
Phase: 180
Phase: 270
Figure 3-4. Four vibratory patterns that differ in phase. Shown above are vibratory patterns with phases of 0 0, 90 0,
180 0, and 270 0.
The Physics of Sound 5
Springs and Masses
We have noted that objects can vibrate at different frequencies, but so far have not discussed the physical
characteristics that are responsible for variations in frequency. There are many factors that affect the natural
vibrating frequency of an object, but among the most important are the mass and stiffness of the object. The effects
of mass and stiffness on natural vibrating frequency can be illustrated with the simple spring-and-mass systems
shown in Figure 3-6. In the pair of spring-and-mass systems to the left, the masses are identical but one spring is
stiffer than the other. If these two spring-and-mass systems are set into vibration, the system with the stiffer spring
will vibrate at a higher frequency than the system with the looser spring. This effect is similar to the changes in
Time ->
Instantaneous Amplitude
0
90
180
270
0/360
Figure 3-5. The system for representing phase treats one cycle of the vibratory pattern as a circle, consisting of 360 0
. A pattern that begins at zero amplitude heading toward positive values (i.e., heading upward) is designated 0 0 phase; a waveform that begins at maximum positive displacement and shows
its initial movement downward has a phase of 90 o
; a waveform that begins at zero and heads

downward has a phase of 180 o; and a waveform that begins at maximum negative displacement and
shows its initial movement upward has a phase of 270 o. . The four phase angles that are shown above
are just examples. An infinite variety of phase angles are possible.
Figure 3-6. A spring and mass system whose natural vibrating frequency is controlled by two parameters: (1) the stiffness of the spring (the stiffer the spring the higher the natural vibrating frequency), and (2) the mass of the material that is suspended from the spring (the greater the mass, the
lower the natural vibrating frequency).

frequency that occur when a guitarist turns the tuning key clockwise or counterclockwise to tune a guitar string by
altering its stiffness.1
The spring-and-mass systems to the right have identical springs but different masses. When these systems are
set into vibration, the system with the greater mass will show a lower natural vibrating frequency. The reason is that
the larger mass shows greater inertia and, consequently, shows greater opposition to changes in direction. Anyone
who has tried to push a car out of mud or snow by rocking it back and forth knows that this is much easier with a
light car than a heavy car. The reason is that the more massive car shows greater opposition to changes in direction.
In summary, the natural vibrating frequency of a spring-and-mass system is controlled by mass and stiffness.
Frequency is directly proportional to stiffness (S↑F↑) and inversely proportional to mass (M↑F↓). It is important to
recognize that these rules apply to all objects, and not just simple spring-and-mass systems. For example, we will
see that the frequency of vibration of the vocal folds is controlled to a very large extent by muscular forces that act
to alter the mass and stiffness of the folds. We will also see that the frequency analysis that is carried out by the
inner ear depends to a large extent on a tuned membrane whose stiffness varies systematically from one end of the
cochlea to the other.
Sound Propagation
As was mentioned at the beginning of this chapter, the generation of a sound wave requires not only vibration,

but also an elastic medium in which the disturbance created by that vibration can be transmitted (see Box 3-1 [bell
jar experiment described in Patrick's science book - not yet written]). To say that air is an elastic medium means that
air, like all other matter, tends to return to its original shape after it is deformed through the application of a force.
The prototypical example of an object that exhibits this kind of restoring force is a spring. To understand the
mechanism underlying sound propagation, it is useful to think of air as consisting of collection of particles that are
connected to one another by springs, with the springs representing the restoring forces associated with the elasticity
of the medium. Air pressure is related to particle density. When a volume of air is undisturbed, the individual
particles of air distribute themselves more-or-less evenly, and the elastic forces are at their resting state. A volume of
air that is in this undisturbed state it is said to be at atmospheric pressure. For our purposes, atmospheric pressure
can be defined in terms of two interrelated conditions: (1) the air molecules are approximately evenly spaced, and
(2) the elastic forces, represented by the interconnecting springs, are neither compressed nor stretched beyond their
resting state. When a vibratory disturbance causes the air particles to crowd together (i.e., producing an increase in
particle density), air pressure is higher than atmospheric, and the elastic forces are in a compressed state.
Conversely, when particle spacing is relatively large, air pressure is lower than atmospheric.
1The example of tuning a guitar string is imperfect since the mass of the vibrating portion of the string decreases slightly as the string is
tightened. This occurs because a portion of the string is wound onto the tuning key as it is tightened.
a b c d e f g h i
a b c d e f g h i
a b c d e f g h i
a b c d e f g h i
a b c d e f g h i
a b c d e f g h i
a b c d e f g h i
a b c d e f g h i
a b c d e f g h i
a b c d e f g h i

a b c d e f g h i
TIME
Figure 3-7. Shown above is a highly schematic illustration of the chain reaction that results in the propagation of a sound wave (modeled after Denes and Pinson, 1963).

When a vibrating object is placed in an elastic medium, an air pressure disturbance is created through a chain reaction similar to that illustrated in Figure 3-7. As the vibrating object (a tuning fork in this case) moves to the right, particle a, which is immediately adjacent to the tuning fork, is displaced to the right. The elastic force generated between particles a and b (not shown in the figure) has the effect a split second later of displacing particle b to the right. This disturbance will eventually reach particles c, d, e, and so on, and in each case the particles will be momentarily crowded together. This crowding effect is called compression or condensation, and it is characterized by dense particle spacing and, consequently, air pressure that is slightly higher than atmospheric pressure. The propagation of the disturbance is analogous to the chain reaction that occurs when an arrangement of dominos is toppled over. Figure 3-7 also shows that at some close distance to the left of a point of compression, particle spacing will be greater than average, and the elastic forces will be in a stretched state. This effect is called rarefaction, and it is characterized by relatively wide particle spacing and, consequently, air pressure that is slightly lower than atmospheric pressure.

The compression wave, along with the rarefaction wave that immediately follows it, will be propagated outward at the speed of sound. The speed of sound varies depending on the average elasticity and density of the medium in which the sound is propagated, but a good working figure for air is about 35,000 centimeters per second, or approximately 783 miles per hour. Although Figure 3-7 gives a reasonably good idea of how sound propagation works, it is misleading in two respects. First, the scale is inaccurate to an absurd degree: a single cubic inch of air contains approximately 400 billion molecules, and not the handful of particles shown in the figure. Consequently,

the compression and rarefaction effects are statistical rather than strictly deterministic as shown in Figure 3-7.

Second, although Figure 3-7 makes it appear that the air pressure disturbance is propagated in a simple straight line from the vibrating object, it actually travels in all directions from the source. This idea is captured somewhat better in Figure 3-8, which shows sound propagation in two of the three dimensions in which the disturbance will be transmitted. The figure shows rod and piston connected to a wheel spinning at a constant speed. Connected to the piston is a balloon that expands and contracts as the piston moves in and out of the cylinder. As the balloon expands the air particles are compressed; i.e., air pressure is momentarily higher than atmospheric. Conversely, when the balloon contracts the air particles are sucked inward, resulting in rarefaction. The alternating compression and rarefaction waves are propagated outward in all directions form the source. Only two of the three dimensions are shown here; that is, the shape of the pressure disturbance is actually spherical rather than the circular pattern that is shown here. Superimposed on the figure, in the graph labeled “one line of propagation,” is the resulting air pressure waveform. Note that the pressure waveform takes on a high value during instants of compression and a low value during instants of rarefaction. The figure also gives some idea of where the term uniform circular motion comes from. If one were to make a graph plotting the height of the connecting rod on the rotating wheel as a function of time it would trace out a perfect sinusoid; i.e., with exactly the shape of the pressure waveform that is superimposed on the figure.

The Sound Pressure Waveform

Returning to Figure 3-7 for a moment, imagine that we chose some specific distance from the tuning fork to observe how the movement and density of air particles varied with time. We would see individual air particles oscillating small distances back and forth, and if we monitored particle density we would find that high particle density (high air pressure) would be followed a moment later by relatively even particle spacing (atmospheric

pressure), which would be followed by a moment later by wide particle spacing (low air pressure), and so on.

Therefore, for an object that is vibrating sinusoidally, a graph showing variations in instantaneous air pressure over time would also be sinusoidal. This is illustrated in Figure 3-9.

The vibratory patterns that have been discussed so far have all been sinusoidal. The concept of a sinusoid has not been formally defined, but for our purposes it is enough to know that a sinusoid has precisely the smooth shape that is shown in Figures such as 3-4 and 3-5. While sinusoids, also known as pure tones, have a very special place in acoustic theory, they are rarely encountered in nature. The sound produced by a tuning fork comes quite close to a sinusoidal shape, as do the simple tones that are used in hearing tests. Much more common in both speech and music are more complex, nonsinusoidal patterns, to be discussed below. As will be seen in later chapters, these complex vibratory patterns play a very important role in speech.

The Frequency Domain

We now arrive at what is probably the single most important concept for understanding both hearing and speech acoustics. The graphs that we have used up to this point for representing either vibratory motion or the air pressure disturbance created by this motion are called time domain representations. These graphs show how instantaneous displacement (or instantaneous air pressure) varies over time. Another method for representing either sound or vibration is called a frequency domain representation, also known as a spectrum. There are, in fact, two kinds of frequency domain representations that are used to characterize sound. One is called an amplitude spectrum (also known as a magnitude spectrum or a power spectrum, depending on how the level of the signal is represented) and the other is called a phase spectrum. For reasons that will become clear soon, the amplitude spectrum is by far the more important of the two. An amplitude spectrum is simply a graph showing what frequencies are present with what amplitudes. Frequency is given along the x axis and some measure of amplitude is given on the y axis. A phase

spectrum is a graph showing what frequencies are present with what phases.
Figure 3-10 shows examples of the amplitude and phase spectra for several sinusoidal signals. The top panel
shows a time-domain representation of a sinusoid with a period of 10 ms and, consequently, a frequency of 100 Hz
(f = 1/t = 1/0.01 sec = 100 Hz). The peak-to-peak amplitude for this signal is 400 μPa, and the signal has a phase of
90 o. Since the amplitude spectrum is a graph showing what frequencies are present with what amplitudes, the
amplitude spectrum for this signal will show a single line at 100 Hz with a height of 400 μPa. The phase spectrum is
a graph showing what frequencies are present with what phases, so the phase spectrum for this signal will show a
single line at 100 Hz with a height of 90 o
. The second panel in Figure 3-10 shows a 200 Hz sinusoid with a peak-to-
peak amplitude of 200 μPa and a phase of 180 o
. Consequently, the amplitude spectrum will show a single line at 200
Hz with a height of 100 μPa, while the phase spectrum will show a line at 200 Hz with a height of 180 o.

Complex Periodic Sounds

Sinusoids are sometimes referred to as simple periodic signals. The term "periodic" means that there is a
pattern that repeats itself, and the term "simple" means that there is only one frequency component present. This is
confirmed in the frequency domain representations in Figure 3-10, which all show a single frequency component in
both the amplitude and phase spectra. Complex periodic signals involve the repetition of a nonsinusoidal pattern,
and in all cases, complex periodic signals consist of more than a single frequency component. All nonsinusoidal
periodic signals are considered complex periodic.
Figure 3-8 Illustration of the propagation of a sound wave in two dimensions.

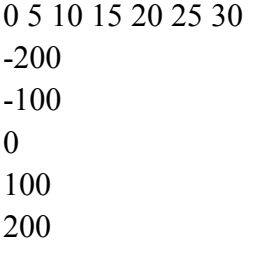
Figure 3-11 shows several examples of complex periodic signals, along with the amplitude spectra for these signals.
The time required to complete one cycle of the complex pattern is called the fundamental period. This is precisely
the same concept as the term period that was introduced earlier. The only reason for using the term "fundamental

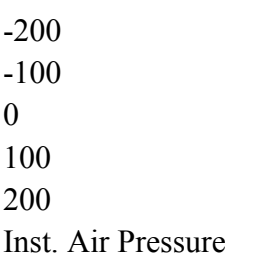
period" instead of the simpler term "period" for complex periodic signals is to differentiate the fundamental period
(the time required to complete one cycle of the pattern as a whole) from other periods that may be present in the
signal (e.g., more rapid oscillations that might be observed within each cycle). The symbol for fundamental period is
t o. Fundamental frequency (f o) is calculated from fundamental period using the same kind of formula that we used
earlier for sinusoids:

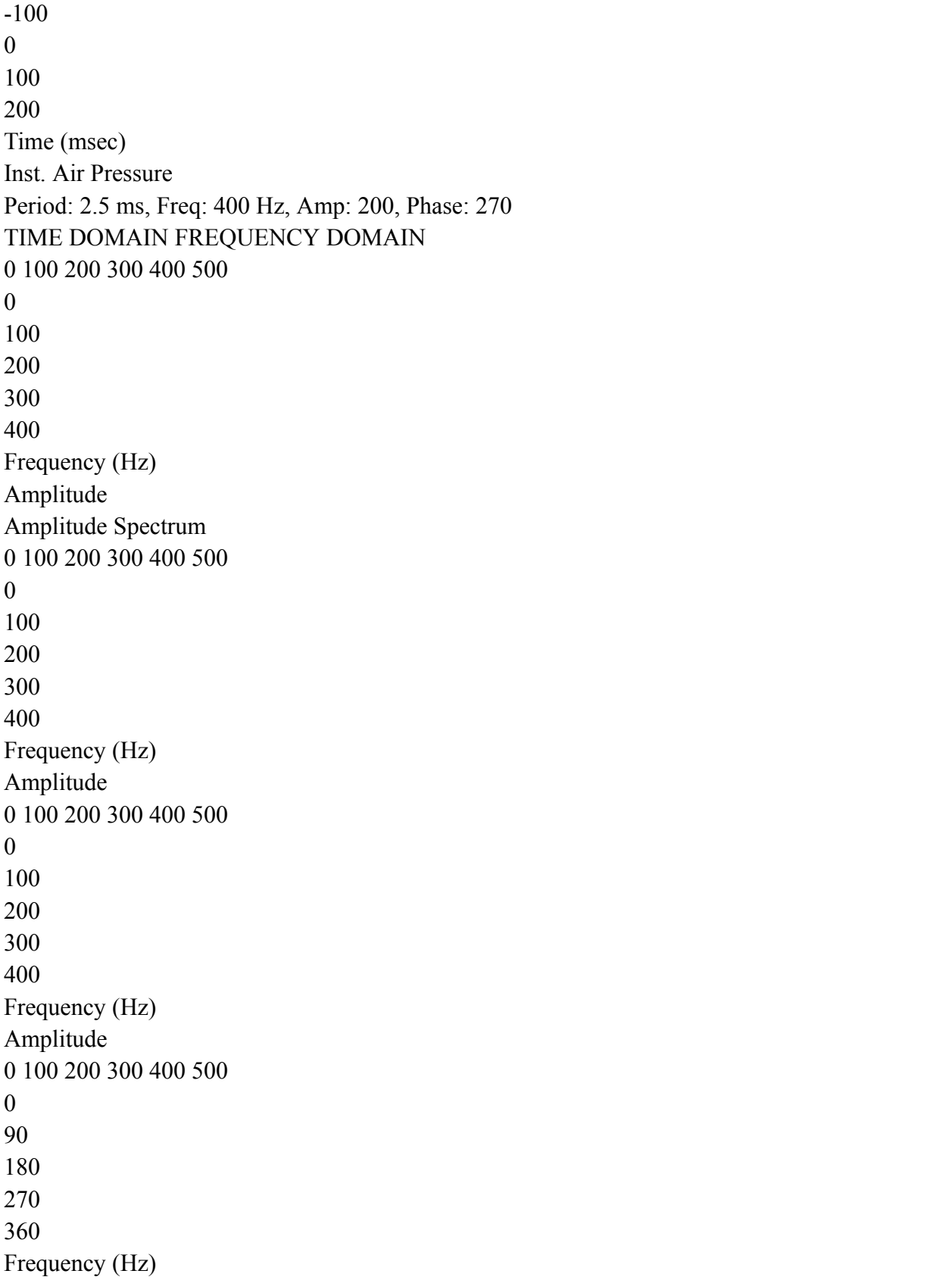
$$f_o = 1/t_o$$

The signal in the top panel of Figure 3-11 has a fundamental period of 5 ms, so fo = 1/0.005 = 200 Hz.

Examination of the amplitude spectra of the signals in Figure 3-11 confirms that they do, in fact, consist of
more than a single frequency. In fact, complex periodic signals show a very particular kind of amplitude spectrum
called a harmonic spectrum. A harmonic spectrum shows energy at the fundamental frequency and at whole
number multiples of the fundamental frequency. For example, the signal in the top panel of Figure 3-11 has energy
present at 200 Hz, 400 Hz, 600 Hz, 800 Hz, 1,000 Hz, 1200 Hz, and so on. Each frequency component in the






-100
0
100
200
Time (msec)
Inst. Air Pressure
Period: 2.5 ms, Freq: 400 Hz, Amp: 200, Phase: 270
TIME DOMAIN FREQUENCY DOMAIN
0 100 200 300 400 500
0
100
200
300
400
Frequency (Hz)
Amplitude
Amplitude Spectrum
0 100 200 300 400 500
0
100
200
300
400
Frequency (Hz)
Amplitude
0 100 200 300 400 500
0
100
200
300
400
Frequency (Hz)
Amplitude
0 100 200 300 400 500
0
90
180
270
360
Frequency (Hz)

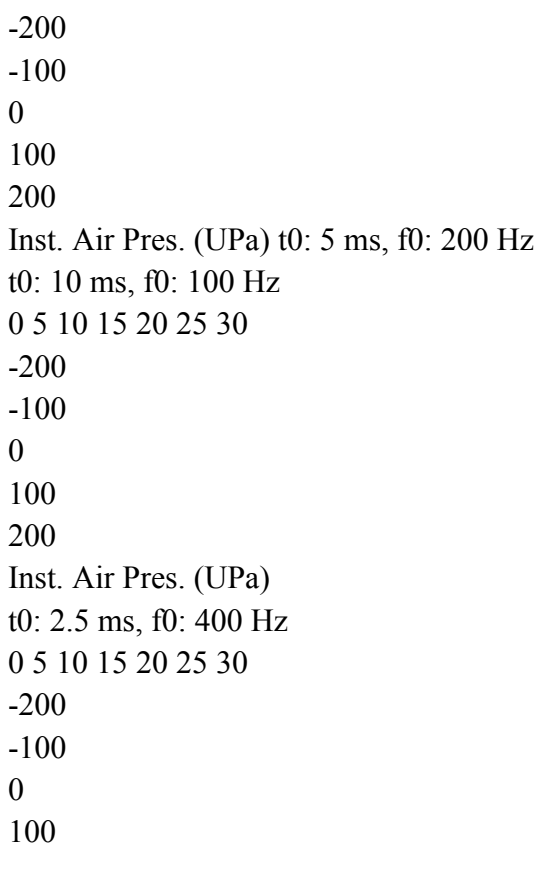
Phase
Phase Spectrum
0 100 200 300 400 500
0
90
180
270
360
Frequency (Hz)
Phase
0 100 200 300 400 500
0
90
180
270
360
Frequency (Hz)
Phase
Figure 3-10. Time and frequency domain representations of three sinusoids. The frequency domain
consists of two graphs: an amplitude spectrum and a phase spectrum. An amplitude spectrum is a graph showing what frequencies are present with what amplitudes, and a phase spectrum is a graph
showing the phases of each frequency component.

amplitude spectrum of a complex periodic signal is called a harmonic (also known as a partial). The fundamental
frequency, in this case 200 Hz, is also called the first harmonic, the 400 Hz component (2 · fo) is called the second
harmonic, the 600 Hz component (3 · fo) is called the third harmonic, and so on.

The second panel in Figure 3-11 shows a complex periodic signal with a fundamental period of 10 ms and,
consequently, a fundamental frequency of 100 Hz. The harmonic spectrum that is associated with this signal will
therefore show energy at 100 Hz, 200 Hz, 300 Hz, 400 Hz, 500 Hz, and so on. The bottom panel of Figure 3-11
shows a complex periodic signal with a fundamental period of 2.5 ms, a fundamental frequency of 400 Hz, and
harmonics at 400, 800, 1200, 1600, and so on. Notice that there two completely interchangeable ways to define the

term fundamental frequency. In the time domain, the fundamental frequency is the number of cycles of the complex
pattern that are completed in one second. In the frequency domain, except in the case of certain special signals, the
fundamental frequency is the lowest harmonic in the harmonic spectrum. Also, the fundamental frequency defines
the harmonic spacing; that is, when the fundamental frequency is 100 Hz, harmonics will be spaced at 100 Hz

Figure 3-11. Time and frequency domain representations of three complex periodic signals. Complex periodic signals have harmonic spectra, with energy at the fundamental frequency (f0) and
at whole number multiples of f0 (f0. 2, f0. 3, f0. 4, etc.) For example, the signal in the upper left, with a
fundamental frequency of 200 Hz, shows energy at 200 Hz, 400 Hz, 600 Hz, etc. In the spectra on
the right, amplitude is measured in arbitrary units. The main point being made in this figure is the
distribution of harmonic frequencies at whole number multiples of f0 for complex periodic signals.



200
Time (msec)
Inst. Air Pres. (UPa)
0 200 400 600 800 1000 1200 1400 1600
0
20
40
60
80
100
120
Frequency (Hz)
Amplitude
0 200 400 600 800 1000 1200 1400 1600
0
20
40
60
80
100
120
Frequency (Hz)
Amplitude
0 200 400 600 800 1000 1200 1400 1600
0
20
40
60
80
100
120
Frequency (Hz)
Amplitude
TIME DOMAIN FREQUENCY DOMAIN
The Physics of Sound 11
0 10 20 30 40 50
-200
-100
0
100

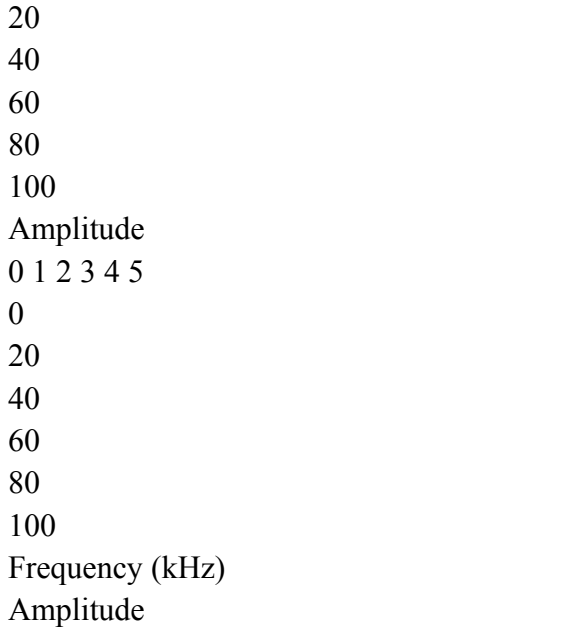
200
Inst. Air Pres. (UPa)
White Noise
/s/
0 10 20 30 40 50
-200
-100
0
100
200
Inst. Air Pres. (UPa)
/f/
0 10 20 30 40 50
-200
-100
0
100
200
TIME (msec)
Inst. Air Pres. (UPa)
0 1 2 3 4 5 6 7 8 9 10
0
20
40
60
80
100
Amplitude
0 1 2 3 4 5 6 7 8 9 10
0
20
40
60
80
100
Amplitude
0 1 2 3 4 5 6 7 8 9 10
0
20
40

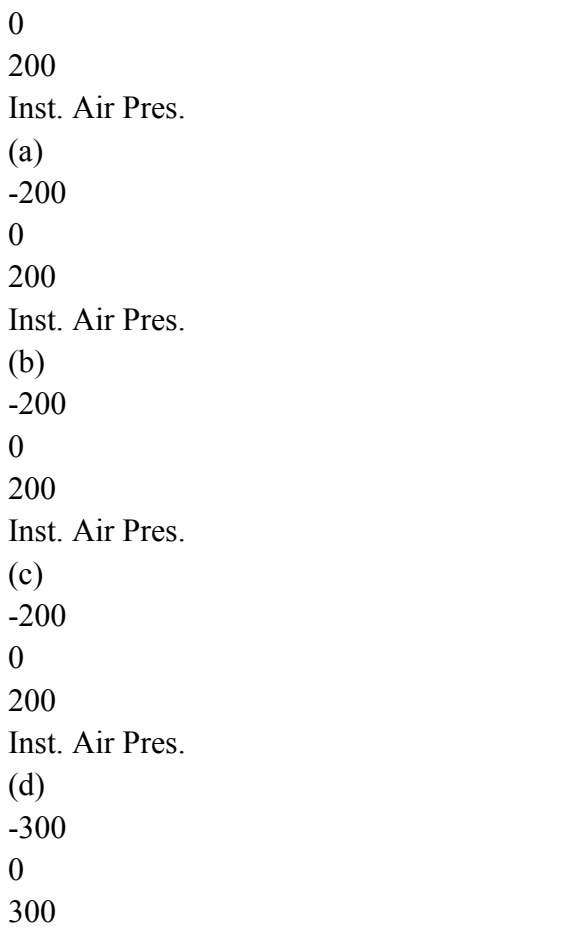
60
80
100
Frequency (kHz)
Amplitude
TIME DOMAIN FREQUENCY DOMAIN
intervals (i.e., 100, 200, 300 ...), when the fundamental frequency is 125 Hz, harmonics will be spaced at 125 Hz
intervals (i.e., 125, 250, 375...), and when the fundamental frequency is 200 Hz, harmonics will be spaced at 200 Hz
intervals (i.e., 200, 400, 600 ...). (For some special signals this will not be the case.2) So, when fo is low, harmonics
will be closely spaced, and when fo is high, harmonics will be widely spaced. This is clearly seen in Figure 3-11: the
signal with the lowest f0 (100 Hz, the middle signal) shows the narrowest harmonic spacing, while the signal with
the highest f0 (400 Hz, the bottom signal) shows the widest harmonic spacing.
There are certain characteristics of the spectra of complex periodic sounds that can be determined by making simple
measurements of the time domain signal, and there are certain other characteristics that require a more complex
analysis. For example, simply by examining the signal in the bottom panel of Figure 3-11 we can determine that it is
complex periodic (i.e., it is periodic but not sinusoidal) and therefore it will show a harmonic spectrum with energy
at whole number multiples of the fundamental frequency. Further, by measuring the fundamental period (2.5 ms)
2There are some complex periodic signals that have energy at odd multiples of the fundamental frequency only. A square wave, for
example, is a signal that alternates between maximum positive amplitude and maximum negative amplitude. The spectrum of square wave shows
energy at odd multiples of the fundamental frequency only. Also, a variety of simple signal processing tricks can be used to create signals with
harmonics at any arbitrary set of frequencies. For example, it is a simple matter to create a signal with energy at 400, 500, and 600 Hz only.
While these kinds of signals can be quite useful for conducting auditory perception experiments, it remains true that most naturally occurring
complex periodic signals have energy at all whole number multiples of the fundamental frequency.

Figure 3-12. Time and frequency domain representations of three non-transient complex aperiodic
signals. Unlike complex periodic signals, complex aperiodic signals show energy that is spread across the spectrum. This type of spectrum is called dense or continuous. These spectra have a very
different appearance from the “picket fence” look that is associated with the discrete, harmonic spectra of complex periodic signals.

and converting it into fundamental frequency (400 Hz), we are able to determine that the signal will have energy at
400, 800, 1200, 1600, etc. But how do we know the amplitude of each of these frequency components? And how do
we know the phase of each component? The answer is that you cannot determine harmonic amplitudes or phases
simply by inspecting the signal or by making simple measurements of the time domain signals with a ruler. We will
see soon that a technique called Fourier analysis is able to determine both the amplitude spectrum and the phase
spectrum of any signal. We will also see that the inner ears of humans and many other animals have developed a
trick that is able to produce a neural representation that is comparable in some respects to an amplitude spectrum.
We will also see that the ear has no comparable trick for deriving a representation that is equivalent to a phase
spectrum. This explains why the amplitude spectrum is far more important for speech and hearing applications than
the phase spectrum. We will return to this point later.
To summarize: (1) a complex periodic signal is any periodic signal that is not sinusoidal, (2) complex periodic
signals have energy at the fundamental frequency (fo) and at whole number multiples of the fundamental frequency
($2 \cdot fo$, $3 \cdot fo$, $4 \cdot fo$ ...), and (3) although measuring the fundamental frequency allows us to determine the frequency
locations of harmonics, there is no simple measurement that can tell us harmonic amplitudes or phases. For this,
Fourier analysis or some other spectrum analysis technique is needed.
Figure 3-13. Time and frequency domain representations of three transients. Transients are complex
aperiodic signals that are defined by their brief duration. Pops, clicks, and the sound gun fire are

examples of transients. In common with longer duration complex aperiodic signals, transients show
dense or continuous spectra, very unlike the discrete, harmonic spectra associated with complex periodic
d
0 10 20 30 40 50 60 70 80 90 100
-200
-100
0
100
200
Inst. Amp. (UPa)
Rap on Desk
Clap
0 10 20 30 40 50 60 70 80 90 100
-200
-100
0
100
200
Inst. Amp. (UPa)
Tap on Cheek
0 10 20 30 40 50 60 70 80 90 100
-200
-100
0
100
200
TIME (msec)
Inst. Amp. (UPa)
0 1 2 3 4 5
0
20
40
60
80
100
Amplitude
0 1 2 3 4 5
0



The Physics of Sound 13


Inst. Air Pres.
(e)
Time ->
Aperiodic Sounds
An aperiodic sound is any sound that does not show a repeating pattern in its time domain representation. There are
many aperiodic sounds in speech. Examples include the hissy sounds associated with fricatives such as /f/ and /s/,
and the various hisses and pops associated with articulatory release for the stop consonants /b,d,g,p,t,k/. Examples of
non-speech aperiodic sounds include a drummer's cymbal or snare drum, the hiss produced by a radiator, and static
sound produced by a poorly tuned radio. There are two types of aperiodic sounds: (1) continuous aperiodic sounds
(also known as noise) and (2) transients. Although there is no sharp cutoff, the distinction between continuous
aperiodic sounds and transients is based on duration. Transients (also "pops" and "clicks") are defined by their very
brief duration, and continuous aperiodic sounds are of longer duration. Figure 3-12 shows several examples of time
domain representations and amplitude spectra for continuous aperiodic sounds. The lack of periodicity in the time
Figure 3-14. Illustration of the principle underlying Fourier analysis. The complex periodic signal
shown in panel e was derived by point-for-point summation of the sinusoidal signals shown in panels a-d. Point-for-point summation simply means beginning at time zero (i.e., the start of the signal) and adding the instantaneous amplitude of signal a to the instantaneous amplitude of signal b
at time zero, then adding that sum to the instantaneous amplitude of signal c, also at time zero, then
adding that sum to instantaneous amplitude of signal d at time zero. The sum of instantaneous amplitudes at time zero of signals a-d is the instantaneous amplitude of the composite signal e at time zero. For example, at time zero the amplitudes of sinusoids a-d are 0, +100, -200, and 0, respectively, producing a sum of -100. This agrees with the instantaneous amplitude at the very beginning of composite signal e. The same summation procedure is followed for all time points.

domain is quite evident; that is, unlike the periodic sounds we have seen, there is no pattern that repeats itself over
time.

All aperiodic sounds -- both continuous and transient -- are complex in the sense that they always consist of energy at more than one frequency. The characteristic feature of aperiodic sounds in the frequency domain is a dense or continuous spectrum, which stands in contrast to the harmonic spectrum that is associated with complex periodic sounds. In a harmonic spectrum, there is energy at the fundamental frequency, followed by a gap with little or no energy, followed by energy at the second harmonic, followed by another gap, and so on. The spectra of aperiodic sounds do not share this "picket fence" appearance. Instead, energy is smeared more-or-less continuously across the spectrum. The top panel in Figure 3-12 shows a specific type of continuous aperiodic sound called white noise. By analogy to white light, white noise has a flat amplitude spectrum; that is, approximately equal amplitude at all frequencies. The middle panel in Figure 3-12 shows the sound /s/, and the bottom panel shows sound /f/. Notice that the spectra for all three sounds are dense; that is, they do not show the "picket fence" look that reveals harmonic structure. As was the case for complex periodic sounds, there is no way to tell how much energy there will be at different frequencies by inspecting the time domain signal or by making any simple measures with a ruler. Likewise, there is no simple way to determine the phase spectrum. So, after inspecting a time-domain signal and determining that it is aperiodic, all we know for sure is that it will have a dense spectrum rather than a harmonic spectrum.

Figure 3-13 shows time domain representations and amplitude spectra for three transients. The transient in the top panel was produced by rapping on a wooden desk, the second is a single clap of the hands, and the third was produced by holding the mouth in position for the vowel /o/, and tapping the cheek with an index finger. Note the brief durations of the signals. Also, as with continuous aperiodic sounds, the spectra associated with transients are dense; that is, there is no evidence of harmonic organization. In speech, transients occur at the instant of articulatory release for stop consonants. There are also some languages, such as the South African languages Zulu, Hottentot,

and Xhosa, that contain mouth clicks as part of their phonemic inventory (MacKay, 1986).

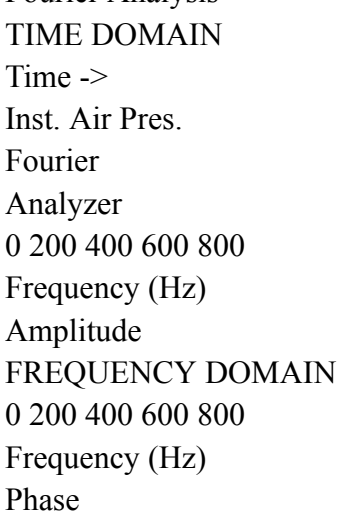


Figure 3-15. A signal enters a Fourier analyzer in the time domain and exits in the frequency domain.

As outputs, the Fourier analyzer produces two frequency-domain representations: an amplitude spectrum that shows the amplitude of each sinusoidal component that is present in the input signal, and a phase spectrum that shows the phase of each of the sinusoids. The input signal can be reconstructed perfectly by summing sinusoids at frequencies, amplitudes, and phase that are shown in the Fourier amplitude and phase spectra, using the summing method that is illustrated in Figure 3-14..

Fourier analysis is an extremely powerful tool that has widespread applications in nearly every major branch of physics and engineering. The method was developed by the 19 th century mathematician Joseph Fourier, and although Fourier was studying thermal waves at the time, the technique can be applied to the frequency analysis of any kind of wave. Fourier's great insight was the discovery that all complex waves can be derived by adding sinusoids together, so long as the sinusoids are of the appropriate frequencies, amplitudes, and phases. For example, the complex periodic signal at the bottom of Figure 3-14 can be derived by summing sinusoids at 100, 200, 300, and 400 Hz, with each sinusoidal component having the amplitude and phase that is shown in the figure (see the caption

of Figure 3-14 for an explanation of what is meant by summing the sinusoidal components). The assumption that all
complex waves can be derived by adding sinusoids together is called Fourier's theorem, and the analysis technique
that Fourier developed from this theorem is called Fourier analysis. Fourier analysis is a mathematical technique that
takes a time domain signal as its input and determines: (1) the amplitude of each sinusoidal component that is
present in the input signal, and (2) the phase of each sinusoidal component that is present in the input signal.
Another way of stating this is that Fourier analysis takes a time domain signal as its input and produces two
frequency domain representations as output: (1) an amplitude spectrum, and (2) a phase spectrum.
The basic concept is illustrated in Figure 3-15, which shows a time domain signal entering the Fourier analyzer.
Emerging at the output of the Fourier analyzer is an amplitude spectrum (a graph showing the amplitude of each
sinusoid that is present in the input signal) and a phase spectrum (a graph showing the phase of each sinusoid that is
present in the input signal). The amplitude spectrum tells us that the input signal contains: (1) 200 Hz sinusoid with
an amplitude of 100 μPa, a 400 Hz sinusoid with an amplitude of 200 μPa, and a 600 Hz sinusoid with an amplitude
of 50 μPa. Similarly, the phase spectrum tells us that the 200 Hz sinusoid has a phase of 90 o, the 400 Hz sinusoid
has a phase of 180 o, and the 600 Hz sinusoid has a phase of 270 o. If Fourier's theorem is correct, we should be able
to reconstruct the input signal by summing sinusoids at 200, 400, and 600 Hz, using the amplitudes and phases that
are shown. In fact, summing these three sinusoids in this way would precisely reproduce the original time domain
signal; that is, we would get back an exact replica of our original signal, and not just a rough approximation to it.
For our purposes it is not important to understand how Fourier analysis works. The most important point about
Fourier's idea is that, visual appearances aside, all complex waves consist of sinusoids of varying frequencies,
amplitudes, and phases. In fact, Fourier analysis applies not only to periodic signals such as those shown in Figure

3-15, but also to noise and transients. In fact, the amplitude spectra of the aperiodic signals shown in Figure 3-13
were calculated using Fourier analysis. In later chapters we will see that the auditory system is able to derive a
neural representation that is roughly comparable to a Fourier amplitude spectrum. However, as was mentioned
earlier, the auditory system does not derive a representation comparable to a Fourier phase spectrum. As a result,
listeners are very sensitive to changes in the amplitude spectrum but are relatively insensitive to changes in phase.

Some Additional Terminology

Overtones vs. Harmonics: The term overtone and the term harmonic refer to the same concept; they are just
counted differently. As we have seen, in a harmonic series such as 100, 200, 300, 400, etc., the 100 Hz component
can be referred to as either the fundamental frequency or the first harmonic; the 200 Hz component is the second
harmonic, the 300 Hz component is the third harmonic, and so on. An alternative set of terminology would refer to
the 100 Hz component as the fundamental frequency, the 200 Hz component as the first overtone, the 300 Hz
component as the second overtone, and so on. Use of the term overtone tends to be favored by those interested in
musical acoustics, while most other acousticians tend to use the term harmonic.

Octaves vs. Harmonics: An octave refers to a doubling of frequency. So, if we begin at 100 Hz, the next octave up
would 200 Hz, the next would be 400 Hz, the next would be 800 Hz, and so on. Note that this is quite different from
a harmonic progression. A harmonic progression beginning at 300 Hz would be 300, 600, 900, 1200, 1500, etc.,
while an octave progression would be 300, 600, 1200, 2400, 4800, etc. There is something auditorilly natural about
octave spacing, and octaves play a very important role in the organization of musical scales. For example, on a piano
keyboard, middle A (A5) is 440 Hz, A above middle A (A 6) is 880 Hz, A7 is 1,760 and so on. (See Box 3-2).

Wavelength: The concept of wavelength is best illustrated with an example given by Small (1973). Small asks us
to imagine dipping a finger repeatedly into a puddle of water at a perfectly regular interval. Each time the finger hits

the water, a wave is propagated outward, and we would see a pattern formed consisting of a series of concentric

circles (see Figure 3-16). Wavelength is simply the distance between the adjacent waves. Precisely the same concept
can be applied to sound waves: wavelength is simply the distance between one compression wave and the next (or
one rarefaction wave and the next or, more generally, the distance between any two corresponding points in adjacent
waves). For our purposes, the most important point to be made about wavelength is that there is a simple
relationship between frequency and wavelength. Using the puddle example, imagine that we begin by dipping our
finger into the puddle at a very slow rate; that is, with a low "dipping frequency." Since the waves have a long
period of time to travel from one dip to the next, the wavelength will be large. By the same reasoning, the
wavelength becomes smaller as the "dipping frequency" is increased; that is, the time allowed for the wave to travel
at high "dipping frequency" is small, so the wavelength is small. Wavelength is a measure of distance, and the
formula for calculating wavelength is a straightforward algebraic rearrangement of the familiar "distance = rate ·
time" formula from junior high school.

$\lambda = c/f$, where: λ = wavelength

c = the speed of sound

f = frequency

By rearranging the formula, frequency can be calculated if wavelength and the speed of sound are known:

$f = c/\lambda$

Lower Frequency
(Longer Wavelength)
Higher Frequency
(Shorter Wavelength)

Figure 3-16. Wavelength is a measure of the distance between the crest of one cycle of a wave and the
crest of the next cycle (or trough to trough or, in fact, the distance between any two corresponding
points in the wave). Wavelength and frequency are related to one another. Because the wave has only a

short time to travel from one cycle to the next, high frequencies produce short wavelengths. Conversely, because of the longer travel times, low frequencies produce long wavelengths.

Spectrum Envelope: The term spectrum envelope refers to an imaginary smooth line drawn to enclose an amplitude spectrum. Figure 3-17 shows several examples. This is a rather simple concept that will play a very important role in understanding certain aspects of auditory perception. For example, we will see that our perception of a perceptual attribute called timbre (also called sound quality) is controlled primarily by the shape of the spectrum envelope, and not by the fine details of the amplitude spectrum. The examples in Figure 3-17 show how differences in spectrum envelope play a role in signaling differences in one specific example of timbre called vowel quality (i.e., whether a vowel sounds like /i/ vs. /a/ vs. /u/, etc.). For example, panels a and b in Figure 3-17 show the vowel /å/ produced at two different fundamental frequencies. (We know that the fundamental frequencies are different because one spectrum shows wide harmonic spacing and the other shows narrow harmonic spacing.)

The fact that the two vowels are heard as /a/ despite the difference in fundamental frequency can be attributed to the fact that these two signals have similar spectrum envelopes. Panels c and d in Figure 3-17 show the spectra of two signals with different spectrum envelopes but the same fundamental frequency (i.e., with the same harmonic spacing). As we will see in the chapter on auditory perception, differences in fundamental frequency are perceived as differences in pitch. So, for signals (a) and (b) in Figure 3-17, the listener will hear the same vowel produced at two different pitches. Conversely, for signals (c) and (d) in Figure 3-17, the listener will hear two different vowels produced at the same pitch. We will return to the concept of spectrum envelope in the chapter on auditory perception.

Amplitude Envelope: The term amplitude envelope refers to an imaginary smooth line that is drawn on top of a time domain signal. Figure 3-18 shows sinusoids that are identical except for their amplitude envelopes. It can be

seen that the different amplitude envelopes reflect differences in the way the sounds are turned on and off. For
example, panel a shows a signal that is turned on abruptly and turned off abruptly; panel b shows a signal that is
turned on gradually and turned off abruptly; and so on. Differences in amplitude envelope have an important effect
on the quality of a sound. As we will see in the chapter on auditory perception, amplitude envelope, along with
spectrum envelope discussed above, is another physical parameter that affects timbre or sound quality. For
0 1 2 3
0
10
20
30
40
50
60
70
Frequency (kHz)
Amplitude
(a) Vowel: /a/, f0: 100 Hz
0 1 2 3
0
10
20
30
40
50
60
70
Frequency (kHz)
Amplitude
(b) Vowel: /a/, f0: 200 Hz
0 1 2 3
0
10
20
30
40

50
60
70
Frequency (kHz)
Amplitude
(c)
Vowel: /i/, f0: 150 Hz
0 1 2 3
0
10
20
30
40
50
60
70
Frequency (kHz)
Amplitude
(d)
Vowel: /u/, f0: 150 Hz
Figure 3-17. A spectrum envelope is an imaginary smooth line drawn to enclose an amplitude spectrum. Panels a and b show the spectra of two signals (the vowel /ɑ/) with different fundamental
frequencies (note the differences in harmonic spacing) but very similar spectrum envelopes. Panels c
and d show the spectra of two signals with different spectrum envelopes (the vowels /i/ and /u/ in this
case) but the same fundamental frequencies (i.e., the same harmonic spacing).

example, piano players know that a given note will sound different depending on whether or not the damping pedal
is used. Similarly, notes played on a stringed instrument such as a violin or cello will sound different depending on
whether the note is plucked or bowed. In both cases, the underlying acoustic difference is amplitude envelope.
Acoustic Filters
As will be seen in subsequent chapters, acoustic filtering plays a central role in the processing of sound by the
inner ear. The human vocal tract also serves as an acoustic filter that modifies and shapes the sounds that are created

by the larynx and other articulators. For this reason, it is quite important to understand how acoustic filters work. In
the most general sense, the term filter refers to a device or system that is selective about the kinds of things that are
allowed to pass through versus the kinds of things that are blocked. An oil filter, for example, is designed to allow
oil to pass through while blocking particles of dirt. Of special interest to speech and hearing science are frequency
selective filters. These are devices that allow some frequencies to pass through while blocking or attenuating other
frequencies. (The term attenuate means to weaken or reduce in amplitude).
A simple example of a frequency selective filter from the world of optics is a pair of tinted sunglasses. A piece
of white paper that is viewed through red tinted sunglasses will appear red. Since the original piece of paper is
white, and since we know that white light consists of all of the visible optical frequencies mixed in equal amounts,
the reason that the paper appears red through the red tinted glasses is that optical frequencies other than those
corresponding to red are being blocked or attenuated by the optical filter. As a result, it is primarily the red light that
is being allowed to pass through. (Starting at the lowest optical frequency and going to the highest, light will appear
red, orange, yellow, green, blue, indigo, and violet.)
Inst. Air Pres.
(a)
Signals Differing in Amplitude Envelope
Inst. Air Pres.
(b)
Inst. Air Pres.
(c)
Time ->
Inst. Air Pres.
(d)
Figure 3-18. Amplitude envelope is an imaginary smooth line drawn to enclose a time-domain signal.
This feature describes how a sound is turned on and turned off; for example, whether the sound is
turned on abruptly and turned off abruptly (panel a), turned on gradually and turned off abruptly (panel

b), turned on abruptly and turned off gradually (panel c), or turned on and off gradually (panel d).

A graph called a frequency response curve is used to describe how a frequency selective filter will behave. A frequency response curve is a graph showing how energy at different frequencies will be affected by the filter.

Specifically, a frequency response curve plots a variable called "gain" as a function of variations in the frequency of the input signal. Gain is the amount of amplification provided by the filter at different signal frequencies. Gains are interpreted as amplitude multipliers; for example, suppose that the gain of a filter at 100 Hz is 1.3. If a 100 Hz sinusoid enters the filter measuring 10 uPa, the amplitude at the output of the filter at 100 Hz will measure 13 μPa (10 μPa x 1.3 = 13 μPa). The only catch in this scheme is that gains can and very frequently are less than 1, meaning that the effect of the filter will be to attenuate the signal. For example, if the gain at 100 Hz is 0.5, a 10 μPa input signal at 100 Hz will measure 5 μPa at the output of the filter. When the filter gain is 1.0, the signal is unaffected by the filter; i.e., a 10 μPa input signal will measure 10 μPa at the output of the filter.

Figure 3-19 shows frequency response curves for several optical filters. Panel a shows a frequency response curve for the red optical filter discussed in the example above. If we put white light into the filter in panel a, the signal amplitude at the output of the filter will be high only when the frequency of the input signal is low. This is because the gain of the filter is high only in the low-frequency portion of the frequency-response curve. This is an example of a lowpass filter; that is, a filter that allows low frequencies to pass through. Panel b shows an optical filter that has precisely the reverse effect on an input signal; that is, this filter will allow high frequencies to pass through while attenuating low- and mid-frequency signals. A white surface viewed through this filter would therefore appear violet. This is an example of a highpass filter. Panel c shows the frequency response curve for a filter that allows a band of energy in the center of the spectrum to pass through while attenuating signal components

of higher and lower frequency. A white surface viewed through this filter would appear green. This is called a
bandpass filter.
Acoustic filters do for sound exactly what optical filters do for light; that is, they allow some frequencies to pass
through while attenuating other frequencies. To get a better idea of how a frequency response curve is measured,
imagine that we ask a singer to attempt to shatter a crystal wine glass with a voice signal alone. To see how the
frequency response curve is created we have to make two rather unrealistic assumptions: (1) we need to assume that
the singer is able to produce a series of pure tones of various frequencies (the larynx, in fact, produces a complex
periodic sound and not a sinusoid), and (2) the amplitudes of these pure tones are always exactly the same. The wine
glass will serve as the filter whose frequency response curve we wish to measure. As shown in Figure 3-20, we
attach a vibration meter to the wine glass, and the reading on this meter will serve as our measure of output
Figure 3-19. Frequency response curves for three optical filters. The lowpass filter on the left allows
low frequencies to pass through, while attenuating or blocking optical energy at higher frequencies.
The highpass filter in the middle has the opposite effect, allowing high frequencies to pass through,
while attenuating or blocking optical energy at lower frequencies. The bandpass filter on the right
allows a band of optical frequencies in the center of the spectrum to pass through, while attenuating or
blocking energy at higher and lower frequencies.

amplitude for the filter. For the purpose of this example, will assume that the signal frequency needed to break the
glass is 500 Hz. We now ask the singer to produce a low frequency signal, say 50 Hz. Since this frequency is quite
remote from the 500 Hz needed to break the glass, the output amplitude measured by the vibration meter will be
quite low. As the singer gets closer and closer to the required 500 Hz, the measured output amplitude will increase

systematically until the glass finally breaks. If we assume that the glass does not break but rather reaches a
maximum amplitude just short of that required to shatter the glass, we can continue our measurement of the
frequency response curve by asking the singer to produce signals that are increasingly high in frequency. We would
find that the output amplitude would become lower and lower the further we got from the 500 Hz natural vibrating
frequency of the wine glass. The pattern that is traced by our measures of output amplitude at each signal frequency
would resemble the frequency response curve we saw earlier for green sunglasses; that is, we would see the
frequency response curve for a bandpass filter.

Additional Comments on Filters

Cutoff Frequency, Center Frequency, Bandwidth. The top panel of Figure 3-21 shows frequency response curves
for two lowpass filters that differ in a parameter called cutoff frequency. Both filters allow low frequencies to pass
through while attenuating high frequencies; the filters differ only in the frequency at which the attenuation begins.
The bottom panel of Figure 3-21 shows two highpass filters that differ in cutoff frequency. There are two additional
terms that apply only to bandpass filters. In our wineglass example above, the natural vibrating frequency of the
wine glass was 300 Hz. For this reason, when the frequency response curve is measured, we find that the wine glass
reaches its maximum output amplitude at 300 Hz. This is called the center frequency or resonance of the filter. It
is possible for two bandpass filters to have the same center frequency but differ with respect to a property called

Figure 3-20. Illustration of how the frequency response curve of a crystal wine glass might be measured. Our singer produces a series of sinusoids that are identical in amplitude but cover a wide range of frequencies. (This part of the example is unrealistic: the human larynx produces a complex sound rather than a sinusoid.) The gain of the wine glass filter can be traced out by measuring the amplitude of vibration at the different signal frequencies.)

bandwidth. Figure 3-22 shows two filters that differ in bandwidth. The tall, thin frequency response curve describes

a narrow band filter. For this type of filter, output amplitude reaches a very sharp peak at the center frequency and
drops off abruptly on either side of the peak. The other frequency response curve describes a wide band filter (also
called broad band). For the wide band filter, the peak that occurs at the resonance of the filter is less sharp and the
drop in output amplitude on either side of the center frequency is more gradual.
Fixed vs. Variable Filters. A fixed filter is a filter whose frequency response curve cannot be altered. For example,
an engineer might design a lowpass filter that attenuates at frequencies above 500 Hz, or a bandpass filter that passes
with a center frequency of 1,000 Hz. It is also possible to create a filter whose characteristics can be varied. For
example, the tuning dial on a radio controls the center frequency of a narrow bandpass filter that allows a single
radio channel to pass through while blocking channels at all other frequencies. The human vocal tract is an example
0 1000 2000 3000 4000
0.0
0.2
0.4
0.6
0.8
1.0
Frequency (Hz)
Gain
Lowpass Filters with Different
Cutoff Frequencies
0 1000 2000 3000 4000
0.0
0.2
0.4
0.6
0.8
1.0
Frequency (Hz)
Gain
Highpass Filters with Different
Cutoff Frequencies
Figure 3-21. Lowpass and highpass filters differing in cutoff frequency.

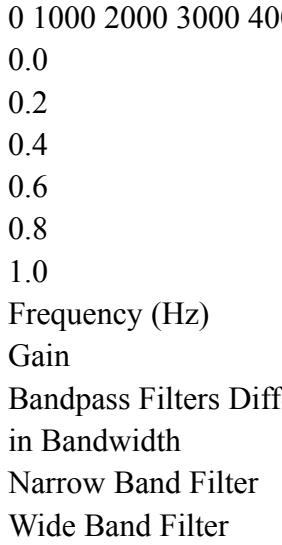


Figure 3-22. Frequency response curves for two bandpass filters with identical center frequencies but different bandwidths. Both filters pass a band of energy centered around 2000 Hz, but the narrow band filter is more selective than the wide band filter; that is, gain decreases at a higher rate above and below the center frequency for the narrow band filter than for the wide band filter

of a variable filter of the most spectacular sort. For example: (1) during the occlusion interval that occurs in the production of a sound like /b/, the vocal tract behaves like a lowpass filter; (2) in the articulatory posture for sounds like /s/ and /sh/ the vocal tract behaves like a highpass filter; and (3) in the production of vowels, the vocal tract behaves like a series of bandpass filters connected to one another, and the center frequencies of these filters can be adjusted by changing the positions of the tongue, lips, and jaw. To a very great extent, the production of speech involves making adjustments to the articulators that have the effect of setting the vocal tract filter in differ modes to produce the desired sound quality. We will have much more to say about this in later chapters.

Frequency Response Curves vs. Amplitude Spectra. It is not uncommon for students to confuse a frequency response curve with an amplitude spectrum. The axis labels are rather similar: an amplitude spectrum plots amplitude on the y axis and frequency on the x axis, while a frequency response curve plots gain on the y axis and frequency on the x axis. The apparent similarities are deceiving, however, since a frequency response curve and an

amplitude spectrum display very different kinds of information. The difference is that an amplitude spectrum
describes a sound while a frequency response curve describes a filter. For any given sound wave, an amplitude
spectrum tells us what frequencies are present with what amplitudes. A frequency response curve, on the other hand,
describes a filter, and for that filter, it tells us what frequencies will be allowed to pass through and what frequencies
will be attenuated. Keeping these two ideas separate will be quite important for understanding the key role played by
filters in both hearing and speech science.

Resonance

The concept of resonance has been alluded to on several occasions but has not been formally defined. The term
resonance is used in two different but very closely related ways. The term resonance refers to: (1) the phenomenon
of forced vibration, and (2) natural vibrating frequency (also resonant frequency or resonance frequency) To
gain an appreciation for both uses of this term, imagine the following experiment. We begin with two identical
tuning forks, each tuned to 435 Hz. Tuning fork A is set into vibration and placed one centimeter from tuning fork
B, but not touching it. If we now hold tuning fork B to a healthy ear, we will find that it is producing a 435 Hz tone
that is faint but quite audible, despite the fact that it was not struck and did not come into physical contact with
tuning fork A. The explanation for this "action-at-a-distance" phenomenon is that the sound wave generated by
tuning fork A forces tuning fork B into vibration; that is, the series of compression and rarefaction waves will
alternately push and pull the tuning fork, resulting in vibration at the frequency being generated by tuning fork A.

The phenomenon of forced vibration is not restricted to this "action-at-a-distance" case. The same effect can be
demonstrated by placing a vibrating tuning fork in contact with a desk or some other hard surface. The intensity of
the signal will increase dramatically because the tuning fork is forcing the desk to vibrate, resulting in a larger
volume of air being compressed and rarefied.3

Returning to our original tuning fork experiment, suppose that we repeat this test using two mismatched tuning forks; for example, tuning fork A with a natural frequency of 256 Hz and tuning fork B with a natural vibrating frequency of 435 Hz. If we repeat the experiment – setting tuning fork A into vibration and holding it one centimeter from tuning fork B – we will find that tuning fork B does not produce an audible tone. The reason is that forced vibration is most efficient when the frequency of the driving force is closest to the natural vibration frequency of the object that is being forced to vibrate. Another way to think about this is that tuning fork B in these experiments is behaving like a filter that is being driven by the signal produced by tuning fork A. Tuning forks, in fact, behave like rather narrow bandpass filters. In the experiment with matched tuning forks, the filter was being driven by a signal frequency corresponding to the peak in the filter's frequency response curve. Consequently, the filter produced a great deal of energy at its output. In the experiment with mismatched tuning forks, the filter is being driven by a signal that is remote from the peak in the filter's frequency response curve, producing a low amplitude output signal.

To summarize, resonance refers to the ability of one vibrating system to force another system into vibration. Further, the amplitude of this forced vibration will be greater as the frequency of the driving force approaches the natural vibrating frequency (resonance) of the system that is being forced into vibration.

3The increase in intensity that would occur as the tuning fork is placed in contact with a hard surface does not mean that additional energy is created. The increase in intensity would be offset by a decrease in the duration of the tone, so the total amount of energy would not increase relative to a freely vibrating tuning fork.

Cavity Resonators

An air-filled cavity exhibits frequency selective properties and should be considered a filter in precisely the way that the tuning forks and wine glasses mentioned above are filters. The human vocal tract is an air-filled cavity that behaves like a filter whose frequency response curve varies depending on the positions of the articulators. Tuning

forks and other simple filters have a single resonant frequency. (Note that we will be using the terms "natural
vibrating frequency" and "resonant frequency" interchangeably.) Cavity resonators, on the other hand, can have an
infinite number of resonant frequencies.
A simple but very important cavity resonator is the uniform tube. This is a tube whose cross-sectional area is
the same (uniform) at all points along its length. A simple water glass is an example of a uniform tube. The method
for determining the resonant frequency pattern for a uniform tube will vary depending on whether the tube is closed
at both ends, open at both ends, or closed at just one end. The configuration that is most directly applicable to
problems in speech and hearing is the uniform tube that is closed at one end and open at the other end. The ear canal,
for example, is approximately uniform in cross-sectional area and is closed medially by the ear drum and open
0.0
0.2
0.4
0.6
0.8
1.0
Gain
500 1500 2500 3500 4500
17.5 cm Uniform Tube
0.0
0.2
0.4
0.6
0.8
1.0
Gain
437.5 1312.5 2187.5 3062.5 3937.5
20 cm Uniform Tube
0 1000 2000 3000 4000 5000
0.0
0.2
0.4
0.6

0.8
1.0
Frequency (Hz)
Gain
583.3 1750.0 2916.7 4083.3 5225.0
15 cm Uniform Tube
Figure 3-23. Frequency response curves for three uniform tubes open at one end and closed at the
other. These kinds of tubes have an infinite number of resonances at odd multiples of the lowest resonance. As the figure shows, shortening the tube shifts all resonances to higher frequencies while
lengthening the tube shifts all resonances to lower frequencies.

laterally. Also, in certain configurations the vocal tract is approximately uniform in cross-sectional area and is
effectively closed from below by the vocal folds and open at the lips. The resonant frequencies for a uniform tube
closed at one end are determined by its length. The lowest resonant frequency (F1) for this kind of tube is given by:
F1 = c/4L, where: c = the speed of sound
L = the length of the tube
For example, for a 17.5 cm tube, F1 = c/4L = 35000/70 = 500 Hz. This tube will also have an infinite number of
higher frequency resonances at odd multiples of the lowest resonance:
F1 = F1 . 1 = 500 Hz
F2 = F1 · 3 = 1,500 Hz
F3 = F1 . 5 = 2,500 Hz
F4 = F1 · 7 = 3,500 Hz
The frequency response curve for this tube for frequencies below 4000 Hz is shown in the solid curve in Figure
3-23. Notice that the frequency response curve shows peaks at 500, 1500, 2500, and 3500 Hz, and valleys in
between these peaks. The frequency response curve, in fact, looks like a number of bandpass filters connected in
series with one another. It is important to appreciate that what we have calculated here is a series of natural vibrating
frequencies of a tube. What this means is that the tube will respond best to forced vibration if the tube is driven by
signals with frequencies at or near 500 Hz, 1500 Hz, 2500 Hz, and so on. Also, the resonant frequencies that were

just calculated should not be confused with harmonics. Harmonics are frequency components that are present in the
amplitude spectra of complex periodic sounds; resonant frequencies are peaks in the frequency response curve of
filters.
We next need to see what will happen to the resonant frequency pattern of the tube when the tube length
changes. If the tube is lengthened to 20 cm:
F1 = c/4L = 35,000/80 = 437.5 Hz
F2 = F1 · 3 = 1,312.5 Hz
F3 = F1 · 5 = 2,187.5 Hz
F4 = F1 · 7 = 3,062.5 Hz
It can be seen that lengthening the tube from 17.5 cm to 20 cm has the effect of shifting all of the resonant
frequencies downward (see Figure 3-23). Similarly, shortening the tube has the effect of shifting all of the resonant
frequencies upward. For example, the resonant frequency pattern for a 15 cm tube would be:
F1 = c/4L = 35,000/60 = 583.3 Hz
F2 = F1 · 3 = 1,750 Hz
F3 = F1 · 5 = 2,916.7 Hz
F4 = F1 · 7 = 4,083.3 Hz
The general rule is quite simple: all else being equal, long tubes have low resonant frequencies and short tubes
have high resonant frequencies. This can be demonstrated easily by blowing into bottles of various lengths. The
longer bottles will produce lower tones than shorter bottles. This effect is also demonstrated every time a water glass
is filled. The increase in the frequency of the sound that is produced as the glass is filled occurs because the
resonating cavity becomes shorter and shorter as more air is displaced by water. This simple rule will be quite
useful. For example, it can be applied directly to the differences that are observed in the acoustic properties of
speech produced by men, women, and children, who have vocal tracts that are quite different in length.
Resonant Frequencies and Formant Frequencies
The term "resonant frequency" refers to natural vibrating frequency or, equivalently, to a peak in a frequency
response curve. For reasons that are entirely historical, if the filter that is being described happens to be a human

vocal tract, the term formant frequency is generally used. So, one typically refers to the formant frequencies of the

vocal tract but to the resonant frequencies of a plastic tube, the body of a guitar, the diaphragm of a loudspeaker, or
most any other type of filter other than the vocal tract. This is unfortunate since it is possible to get the mistaken idea
that formant frequencies and resonant frequencies are different sorts of things. The two terms are, in fact, fully
synonymous.

The Decibel Scale

The final topic that we need to address in this chapter is the representation of signal amplitude using the decibel
scale. The decibel scale is a powerful and immensely flexible scale for representing the amplitude of a sound wave.
The scale can sometimes cause students difficulty because it differs from most other measurement scales in not just
one but two ways. Most of the measurement scales with which we are familiar are absolute and linear. The decibel
scale, however, is relative rather than absolute, and logarithmic rather than linear. Neither of these characteristics is
terribly complicated, but in combination they can make the decibel scale appear far more obscure than it is. We will
examine these features one at a time, and then see how they are put together in building the decibel scale.

Linear vs. Logarithmic Measurement Scales

Most measurement scales are linear. To say that a measurement scale is linear means that it is based on equal
additive distances. This is such a common feature of measurement scales that we do not give it much thought. For
example, on a centigrade (or Fahrenheit) scale for measuring temperature, going from a temperature of 90 o to a
temperature of 91 o involves adding one 1 o. One rather obvious consequence of this simple additivity rule is that the
difference in temperature between 10 o and 11 o is the same as the difference in temperature between 90 o and 91 o.
However, there are scales for which this additivity rule does not apply. One of the best known examples is the
Richter scale that is used for measuring seismic intensity. The difference in seismic intensity between Richter values

of 4.0 and 5.0, 5.0 and 6.0, 6.0 and 7.0 is not some constant amount of seismic intensity, but rather a constant
multiple. Specifically, a 7.0 on the Richter scale indicates an earthquake that is 10 times greater in intensity than an
earthquake that measures 6.0 on the Richter scale. Similarly, an 8.0 on the Richter scale is 10 times greater in
intensity than a 7.0. Whenever jumping from one scale value to the next involves multiplying by a constant rather
than adding a constant, the scale is called logarithmic. (The multiplicative constant need not be 10. See Box 3-2 for
an example of a logarithmic scale – an octave progression – that uses 2 as the constant.) Another way of making the
same point is to note that the values along the Richter scale are exponents rather than ordinary numbers; for
example, a Richter value of 6 indicates a seismic intensity of 10^6, a Richter value of 7 indicates a seismic intensity of
10^7, etc. The Richter values can, of course, just as well be referred to as powers or logarithms since both of these
terms are synonyms for exponent. The decibel scale is an example of a logarithmic scale, meaning that it is based on
equal multiples rather than equal additive distances.

Absolute vs. Relative Measurement Scales

A simple example of a relative measurement scale is the Mach scale that is used by rocket scientists to measure
speed. The Mach scale measures speed not in absolute terms but in relation to the speed of sound. For example, a
missile at Mach 2.0 is traveling at twice the speed of sound, while a missile at Mach 0.9 is traveling at 90% of the
speed of sound. So, the Mach scale does not represent a measured speed (S_m) in absolute terms, but rather,
represents a measured speed in relation to a reference speed (S_m/S_r). The reference that is used for the Mach scale is
the speed of sound, so a measured absolute speed can be converted to a relative speed on the Mach scale by simple
division. For example, taking 783 mph as the speed of sound, 1,200 mph = 1200/783 = Mach 1.53. The decibel scale
also exploits this relative measurement scheme. The decibel scale does not represent a measured intensity (I_m) in
absolute terms, but rather, represents the ratio of a measured intensity to a reference intensity (I_m/I_r).

The decibel scale is trickier than the Mach scale in one important respect. For the Mach scale, the reference is
always the speed of sound, but for the decibel scale, many different references can be used. In explaining how the
decibel scale works, we will begin with the commonly used intensity reference of 10 -12 w/m2 (watts per square
meter), which is approximately the intensity that is required for an average normal hearing listener to barely detect a
1,000 Hz pure tone. So, for our initial pass through the decibel scale, 10 -12 w/m2 will serve as I r , and will perform the
same function that the speed of sound does for the Mach scale. Table 3-1 lists several sounds that cover a very broad
range of intensities. The second column shows the measured intensities of those sounds, and the third column shows
the ratio of those intensities to our reference intensity. Whispered speech, for example, measures approximately 10 -8

w/m2, which is 10,000 times more intense than the reference intensity (10 -8 /10 -12 = 10 4 = 10,000). The main point to
be made about column 3 is that the ratios become very large very soon. Even a moderately intense sound like
conversational speech is 1,000,000 times more intense than the reference intensity. The awkwardness of dealing
with these very large ratios has a very simple solution. Column 4 shows the ratios written in exponential notation,
and column 5 simplifies the situation even further by recording the exponent only. The term exponent and the term
logarithm are synonymous, so the measurement scheme that is expressed by the numbers in column 5 can be
summarized as follows: (1) divide a measured intensity by a reference intensity (in this case, 10 -12 w/m2), (2) take the
logarithm of this ratio (i.e., write the number in exponential notation and keep the exponent only). This method, in
fact, is a completely legitimate way to represent signal intensity. The unit of measure is called the bel, after A.G.
Bell, and the formula is:
bel = log10 I m/I r , where: I m = a measured intensity
I r = a reference intensity

Table 3-1. Sound intensities and intensity ratios showing how the decibel scale is created. Column 2 shows the measured intensities (Im) of several sounds. Column 3 shows the ratio of these intensities to a reference intensity of 10 -12 w/m2 . Column 4 shows the ratio written in exponential notation while column 5 shows the exponent only. The last column shows the intensity ratio expressed in decibels, which is simply the logarithm of the intensity ratio multiplied by 10.

Measured Sound Intensity (I m)	Ratio (Im/Ir)	Exponent Exp. Not. (log 10)	Decibel (10 x log 10)
Threshold @ 1 kHz	10 -12 w/m2	1	10 0 0 0 0
Whisper	10 -8 w/m2	10,000	10 4 4 40
Conversational Speech	10 -6 w/m2	1,000,000	10 6 6 60
City Traffic	10 -4 w/m2	100,000,000	10 8 8 80
Rock & Roll	10 -2 w/m2	10,000,000,000	10 10 10 100
Jet Engine	10 0 w/m2	1,000,000,000,000	10 12 12 120

Legitimate or not, the bel finds its sole application in textbooks attempting to explain the decibel. For reasons that are purely historical, the log10 of the intensity ratio is multiplied by 10, changing bel into the decibel (dB). As shown in the last column of Table 3-1, this has the very simple effect of turning 4 bels into 40 decibels, 8 bels into 80 decibels, etc. The formula for the decibel, then, is:

dB IL = 10 log10 I m/I r , where:

I m = a measured intensity

I r = a reference intensity

The designation "IL" stands for intensity level, and it indicates that the underlying measurements are of sound

intensity and not sound pressure. As will be seen below, a different version of this formula is needed if sound
pressure measurements are used. The multiplication by 10 in the dB IL formula is a simple operation, but it can

sometimes have the unfortunate effect of making the formula appear more obscure that it is. The decibel values that
are calculated, however, should be readily interpretable. For example, 30 dB IL means 3 factors of 10 more intense
than I r , 60 dB IL means 6 factors of 10 more intense than I r , and 90 dB IL means 9 factors of 10 more intense than I r .

Deriving a Pressure Version of the dB Formula

In a simple world, we would be finished with the decibel scale. The problem is that the formula is based on
measurements of sound intensity, but as a purely practical matter sound intensity is difficult to measure. Sound
pressure, on the other hand, is quite easy to measure. An ordinary microphone, for example, is a pressure sensitive
device. The problem, then, is that the decibel is defined in terms of intensity measurements, but the measurements
that are actually used will nearly always be measures of sound pressure. This problem can be addressed since there
is a predictable relationship between intensity (I) and pressure (E): intensity is proportional to pressure squared:

I o⊂ E 2

Knowing this relationship allows us to create a completely equivalent version of the decibel formula that will work
when sound pressure measurements are used instead of sound intensity measurements. All we need to do is
substitute squared pressure measurements in place of the intensity measurements:

dB IL = 10 log10 I m/I r (intensity version of formula)

dB SPL= 10 log10 E 2m/E 2 (pressure version of formula)

The designation "SPL" stands for sound pressure level, and it indicates that measures of sound pressure have been
used and not measures of sound intensity. Although the dB SPL formula shown here will work fine, it will almost
never be seen in this form. The reason is that the formula is algebraically rearranged so that the squaring operation is
not needed. The algebra is shown below:

(1) dB IL = 10 log10 I m/I r (the intensity version of the formula)

(2) dB SPL = 10 log10 E 2m/E 2 (measures of E 2 replace measures of I because I o⊂ E 2)
(3) dB SPL = 10 log10 (E m/E r) 2 (a 2
/b 2 = (a/b) 2)
(4) dB SPL = 10 · 2 log10 E m/E r (this is the only tricky step: log a b = b log a)
(5) dB SPL = 20 log10 E m/E r (2 · 10 = 20)
With the possible exception of the fourth step,4 the algebra is straightforward, but the details of the derivation
are less important than the following general points:
1. The decibel formula is defined in terms of intensity ratios. The basic formula is;
dB IL = 10 log10 I m/I r .
2. While sound intensity is difficult to measure, sound pressure is easy to measure. It is therefore necessary to
derive a version of the decibel formula that works when measures of sound pressure are used instead of sound
intensity.
4 Step 4 is the only tricky part of derivation. The reason it works is that squaring a number and then taking a log is the same as taking
the log first, and then multiplying the log by 2. For example, note that the two calculations below produce the same result:
log 10 100 2 = log 10 10,000 = 4 (square first, then take the log)
log 10 100 2 = (log 10 100) x 2 = 2 x 2 = 4 (take the log, then multiply by 2)

3. The derivation of the pressure version of the formula is based entirely on the fact that intensity is proportional to
pressure squared (I o⊂ E 2). This allows measures of E 2 to replace measures of I, turning: dB IL = 10 log10 I m/I r into
dB SPL = 10 log10 E 2m/E r2 . A few algebra tricks are applied to turn this formula into the more aesthetically pleasing
final version: dB SPL = 20 log10 E m/E r .
4. The two versions of the formula are fully equivalent to one another (see Box 3-3).
This last point about the equivalence of the intensity and sound pressure versions of the formula is explained in
some detail in Box 3-3, but the basic point is quite simple. The pressure version of the dB formula was derived from
the intensity version of the formula through algebraic manipulations (based on this relationship: I o⊂ E 2). The whole

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Box 3-2

HARMONICS, OCTAVES, LINEAR SCALES, AND LOGARITHMIC SCALES

As we will see when the decibel scale is introduced, there is an important distinction to be made between linear scales, which are quite common, and logarithmic scales, which are less common but quite important.

This distinction can be illustrated by examining the difference between a harmonic progression and an octave progression. Notice that in a harmonic progression, the spacing between the harmonics is always the same; that is, the difference between H1 and H 2 is the same as the difference between H2 and H 3, and so on. This is because increases in frequency between one harmonic and the next involve adding a constant, with the constant being the fundamental frequency. For example:

H 1 500
H 2 1000 (add 500)
H 3 1500 (add 500)
H 4 2000 (add 500)
. .
. .
. .

To get from one scale value to another on an octave progression involves multiplying by a constant rather than adding a constant. For example, an octave progression starting at 500 Hz looks like this:

O 1 500
O 2 1000 (multiply by 2)
O 3 2000 (multiply by 2)
O 4 4000 (multiply by 2)
. .
. .
. .

As a result of the fact that we are multiplying by a constant rather than adding a constant, the spacing is no longer even (i.e., the spacing between O 1 and O 2 is 500 Hz, the spacing between O2 and O 3 is 1000 Hz, and so on). The point to be made of this is that there are two fundamentally different kinds of scales: (1) scales like harmonic progressions that are created by adding a constant, which are by far the more common, and (2) scales

like octave progressions that are created by multiplying by a constant. Scales that are created by adding a
constant are called linear scales, while scales that are created by multiplying by a constant are called
logarithmic scales. Note that for an octave progression, the multiplier happens to be 2, meaning that progressing
from one frequency to an octave above that frequency involves multiplication by 2. However, a logarithmic
scale can be built using any multiplier. We will return to the distinction between linear and logarithmic scales
when we talk about the decibel scale, and there we will see that a logarithmic scale is built around multiplication
by a constant value of 10 rather than 2.

point of algebra, of course, is to keep the expression on the left equal to the expression on the right. The simple and
useful point that emerges from this is this: If an intensity meter shows that a given sound measures 60 dB IL, for
example, a pressure meter will show that the same sound measures exactly 60 dB SPL. (This may seem
counterintuitive due to the differences in the formulas, but see Box 3-3 for the explanation.) The equivalence of the
two versions of the dB formula greatly simplifies the interpretation of sound levels that are expressed in decibels.

References

The reference that is used for the Mach scale is always the speed of sound. One of the virtues of the decibel
scale is that any reference can be used as long as it is clearly specified. The only reference that has been mentioned
so far is 10 -12 w/m2
, which is roughly the audibility threshold for a 1,000 Hz pure tone. This is a standard reference intensity, and unless otherwise stated it should be assumed that this is used when a signal level is reported in dB IL.

The standard reference that is used for dB SPL is 20 μPa, so when a signal level is reported in dB SPL it should be
assumed that this reference is used unless otherwise stated.5

Many references besides these two standard references can be used. For example, suppose that a speech signal
is presented to a listener at an average level of 3500 μPa in the presence of a noise signal whose average sound
pressure is 1400 μPa. The speech-to-noise ratio (S/N) can be represented on a decibel scale, using the level of the
speech as E m and the level of the noise as E r :

dB s/n = 20 log10 E m/ E r
= 20 log10 3500/1400
= 20 log10 2.5
= 20 (0.39794)
= 7.96 dB

To take one more example, assume that a voice patient prior to treatment produces sustained vowels that
average 2300 μPa. Following treatment the average sound pressures increase to 8890 μPa. The improvement in
sound pressure (post-treatment relative to pre-treatment) can be represented on a decibel scale:

dB Improvement = 20 log10 E post/E pre
= 20 log10 8890/2300
= 20 log10 (3.86522)
= 20 (0.58717)
= 11.74 dB

A final example can be used to make the point that the decibel scale can be used to represent intensity ratios for
any type of energy, not just sound. Bright sunlight has a luminance measuring 100,000 cd/m2 (candela per square
meter). Light from a barely visible star, on the other hand, has a luminance measuring 0.0001 cd/m2. We can now
ask how much more luminous bright sunlight is in relation to barely visible star light, and the dB scale can be used
to represent this value. Since the underlying physical quanities here are measures of electromagnetic intensity, we
want the intensity version of the formula rather than the pressure version.

dB = 10 log10 I sunlight/I starlight
= 10 log10 100000/0.0001
= 10 log10 10 5/10 -4
= 10 log10 10 9 (division is done by subtracting exponents: 5 – (-4) = 9)
= 10 (9)
= 90 dB

5The standard pressure reference for dB SPL is sometimes given as 0.0002 dynes/cm2 rather than 20 μPa. These two sound pressures are
identical, however, in exactly the same sense that 4 quarts and 1 gallon are identical. Likewise, the standard reference for dB IL is often given as
10^{-16} w/cm2 instead of 10^{-12} w/m2
. These two intensities are also identical.

The fact that we are measuring light rather than sound makes no difference: a decibel is $10 \log_{10} I_m/I_r$ (or,
equivalently, $20 \log_{10} E_m/E_r$), regardless of whether the energy comes from sound, light, electrical current, or any
other type of energy.

dB Hearing Level (dB HL)

The dB Hearing Level (dB HL) scale was developed specifically for testing hearing sensitivity for pure tones
of different frequencies. The sound-level dials on clinical audiometers,6 for example, are calibrated in dB HL rather
than dB SPL. To understand the motivation for the dB HL scale examine Figure 3-24, which shows the sound level (in
dB SPL) required for the average, normal-hearing listener to barely detect pure tones at frequencies between 125 and
8000 Hz. This is called the audibility curve and the simple but very important point to notice about this graph is
that the curve is not a flat line; that is, the ear is clearly more sensitive at some frequencies than others. The
differences in sensitivity are quite large in some cases. For example, the average normal-hearing listener will barely
detect a 1000 Hz pure tone at 7 dB SPL, but at 125 Hz the sound level needs to be cranked all the way up to 45 dB SPL,
an increase in intensity of nearly 4000:1. Now suppose we were to test pure-tone sensitivity using an audiometer that
is calibrated in dB SPL. Imagine that a listener barely detects a 1000 Hz pure tone at 25 dB SPL. Does this listener have
a hearing loss, and if so how large? The only way to answer this question is to consult the data in Figure 3-24, which
shows that the threshold of audibility for the average normal hearing listener at 1000 Hz is 7 dB SPL. This means that
the hypothetical listener in this example has a hearing loss of 25-7 = 18 dB. Suppose further that the same listener

detects a 250 Hz tone at 20 dBSPL. The table in Figure 3-24 shows that normal hearing sensitivity at 250 Hz is 25.5
dB SPL, meaning that the listener has slightly better than normal hearing at this frequency. As a final example,
imagine that this listener barely detects a 500 Hz tone at 30 dB SPL. Since the table shows that normal hearing
sensitivity at 500 Hz is 11.5 dB SPL, the listener has a hearing loss of 30.0-11.5 = 18.5 dB. The simple point to be
made about these examples is that, with an audiometer dial that is calibrated in dB SPL, it is not possible to determine
whether a listener has a hearing loss, or to measure the size of that loss, without doing some arithmetic involving the
normative data in Figure 3-24. The dB HL scale, however, provides a simple solution to this problem that avoids this
arithmetic entirely. The solution involves calibrating the audiometer in such a way that, when the level dial is set to
0 dB HL, sound level is set to the threshold of audibility for the average normal-hearing listener for that signal
frequency. For example, when the level dial is set to 0 dB HL at 125 Hz the level of tone will be 45 dB SPL – the
threshold of audibility for the average normal hearing listener at this frequency. Now if a listener barely detects the
125 Hz tone at 0 dB HL, no arithmetic is needed; the listener has normal hearing at this frequency. Further, if the
listener barely detects this 125 Hz tone at 40 dB HL, for example, the listener must have a 40 dB loss at this frequency
– and again it is not necessary to consult the data in Figure 3-24. Similarly, when the level dial is set to 0 dB HL at
250 Hz the level of the tone will be 25.5 dB SPL, which is the audibility threshold at 250 Hz. If this tone is barely
detected at 0 dB HL, the listener has normal hearing at this frequency. However, if the tone is not heard until the dial
is increased to 50 dB dB HL, for example, the listener has a 50 dB hearing loss at this frequency. The same system is
used for all signal frequencies: in all cases, the 0 dB HL reference is not a fixed number as it is for dB SPL (a constant
value of 20 μPa, no matter what the signal frequency is) or dB IL (a constant value of 10 -12 watts/m2, again
independent of signal frequency), but rather a family of numbers. In each case the reference for the dB HL scale is the

threshold of audibility for an average, normal-hearing listener at a particular signal frequency. What this means is
that values in dB HL are a fixed distance above the audibility curve, although they may be very different levels in
dB SPL. For illustration, Figure 3-25 shows the audibility curve (the filled symbols) and, above that in the unfilled
symbols, a collection of values that all measure 30 dB HL. Although the sound levels on the 30 dB HL curve vary
considerably in dB SPL (i.e. measured using 20 μPa as the reference), every data point on this curve is a constant 3
factors of 10, or 30 dB, above the audibility curve. The value of 30 dB in this figure is just an example. All values in
dB HL and dB SPL are interpreted in the same way: 50 dB SPL means that the signal being measured is 100,000 times
(i.e., 5 factors of 10) more intense than the fixed reference of 20 μPa, independent of frequency; 50 dB HL, on the
other hand, means that the signal being measured is 100,000 times (again, 5 factors of 10) more intense than a tone
that is barely audible to a normal-hearing listener at that signal frequency. Similarly, 20 dB SPL means that the signal
is 20 dB (2 factors of 10) more intense than the fixed reference of 20 μPa, while 20 dB HL means that the signal is 20
dB (again, 2 factors of 10) above the audibility curve.
6A clinical audiometer is an instrument with, among other things, one dial (for each ear) that controls pure-tone frequency and another dial
that controls the intensity of the tone. The listener is asked to raise a hand when the tone is barely audible.

Summary
The decibel is a powerful scale for representing signal amplitude. The scale has two important properties: (1)
similar to the Mach scale, it represents signal level not in absolute terms but as a measured level divided by a
reference level; and (2) like the Richter scale, the dB scale is logarithmic rather than linear, meaning that it is based
on equal multiplicative distances rather than equal additive distances. While the decibel is defined in terms of
intensity ratios, for practical reasons, measures of sound pressure are far more common than measures of sound

intensity. Consequently, a version of the decibel formula was derived that makes use of pressure ratios rather than
intensity ratios. The derivation was based on the fact that intensity is proportional to pressure squared. The two
versions of the decibel formula (dB IL = 10 log 10 I m/I r and dB SPL = 20 log 10 E m/E r) are fully equivalent, meaning that
if a sound measures 60 dB IL that same sound will measure 60 dB SPL. Unlike the Mach scale, which always uses the
speed of sound as a reference, any number of references can be used with the decibel scale. The standard reference
for the dB IL scale is 10 -12 w/m2 and the standard reference for the dB SPL scale is 20 μPa. However, any level can be
used as a reference as long as it is specified. The dB HL scale, widely used in audiological assessment, was developed
specifically for measuring sensitivity to pure tones of difference frequencies. The reference that is used for the dB HL
scale is the threshold of audibility at a particular signal frequency for the average, normal-hearing listener. Sound
levels in dB SPL and dB HL are interpreted quite differently. For example, a pure tone measuring 40 dB SPL is 4 factors
of 10 (i.e., 40 dB) greater than the fixed SPL reference of 20 μPa, while a pure tone measuring 40 dB HL is 4 factors
of 10 (again, 40 dB) greater than a tone of that same frequency that is barely audible to an average, normal-hearing
listener.

Frequency	Threshold
125	45.0
250	25.5
500	11.5
750	8.0
1000	7.0
1500	6.5
2000	9.0
3000	10.0
4000	9.5
6000	15.5
8000	13.0

Figure 3-24. The threshold of audibility for the average, normal-hearing listener for pure tones varying between

125 and 8000 Hz. The audibility threshold is the sound level in dBSPL that is required for a listener to barely detect
a tone. Values on this curve are shown in the table to the right. The most important point to note about this graph
is that the curve is not flat, meaning that the ear is more sensitive at some frequencies than others. In particular,
the ear is more sensitive in a range of mid-frequencies between about 1000 and 4000 Hz than it is at lower and
higher frequencies. The complex shape of this curve provides the underlying motivation for the dB HL scale. See
text for details.

Figure 3-25. The lower function is the audibility curve – the sound level in dB SPL that is required for an average
normal hearing listener to barely detect pure tones of different frequencies. The upper function shows sound levels for
a set of tones that all measure 30 dB HL. These tones vary quite a bit in dB SPL (i.e., relative to the constant value of 20
μPa) but in all cases the tones are a constant 3 factors of 10 in intensity (i.e., 30 dB) above the audibility curve.

Box 3-3
THE EQUIVALENCE OF THE INTENSITY AND PRESSURE VERSIONS OF THE DECIBEL FORMULA
One fact about the two versions of the dB formula that is not always well understood is that the dB IL and
dB SPL formulas are fully equivalent. By "fully equivalent" we mean the following: suppose that a sound intensity
meter is used to measure the level of some sound, and we find that this sound is 1,000 times more intense than
the standard intensity reference of 10 -12 w/m2. The sound would then measure 30 dB IL (10 log10 1,000 = 10 (3) =
30 dB IL). Now suppose that we put the sound intensity meter away and use a sound pressure meter to measure
the same sound. You might think that the sound would measure 60 dB SPL since now we are multiplying by 20

instead of 10, but the trick is that the ratio is no longer 1,000. Recall that intensity is proportional to pressure
squared, which means that pressure is proportional to the square root of intensity. This means that if the intensity
ratio is 1,000, the pressure ratio must be the square root of 1,000, or 31.6. So, the formula now becomes 20 log
31.6 = 20 (1.5) = 30 dB SPL, which is exactly what we obtained originally. It will always work out this way: if a
sound measures 50 dB IL, that same sound will measure 50 dB SPL.
Table 3-2 might help to make this more clear. The first column shows an intensity ratio, the second column
shows the corresponding pressure ratio (this is always the square root of the intensity ratio), the third column
shows the dB IL value (10 log of the intensity ratio), and the fourth column shows dB SPL value (20 log of the
pressure ratio). As you can see, they are always the same.

Table 3-2. Intensity ratios, equivalent pressure ratios, dB IL values and dB SPL values showing the equivalence of the intensity and pressure versions of the dB formula.

Intensity Ratio	Pressure Ratio	dB IL (10 log10 I m/I r)	dB SPL (20 log10 E m/E r)
10	3.16	10.00	10.00
20	4.47	13.01	13.01
40	6.32	16.02	16.02
50	7.07	16.99	16.99
60	7.75	17.78	17.78
70	8.37	18.45	18.45
80	8.94	19.03	19.03
90	9.49	19.54	19.54
100	10.00	20.00	20.00
200	14.14	23.01	23.01
300	17.32	24.77	24.77
400	20.00	26.02	26.02
500	22.36	26.99	26.99
1000	31.62	30.00	30.00

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Study Questions: Physical Acoustics

1. Explain the basic processes that are involved in the propagation of a sound wave.
2. Draw time- and frequency-domain representations of simple periodic, complex periodic, complex aperiodic, and transient sounds.
3. Draw time- and frequency-domain representations of two complex periodic sounds with different fundamental frequencies.
4. Draw time-domain representations of two simple periodic sounds with the same frequency and phase, but different amplitudes.
5. Draw time-domain representations of two simple periodic sounds with the same frequency and different amplitudes but different phases.
6. Draw amplitude spectra of two sounds with the same fundamental frequencies but different spectrum envelopes.
7. Draw amplitude spectra of two sounds with different fundamental frequencies but similar spectrum envelopes.
8. Calculate signal frequencies for sinusoids with the following values:

a. period = 0.34 s
b. period = 2 s
c. period = 10 ms
d. period = 2 ms
e. wavelength = 20 cm
f. wavelength = 100 cm

Answers:

a. f = 1/0.34 = 2.94 Hz
b. f = 1/2 = 0.5 Hz
c. f = 1/0.01 = 100 Hz
d. f = 1/.002 = 500 Hz
e. f = c/WL (speed of sound/wavelength) = 35000/20 = 1750 Hz
f. f = c/WL (speed of sound/wavelength) = 35000/100 = 350 Hz

9. Calculate the three lowest resonant frequencies of the following uniform tubes that are closed at one end and open at the other end:

a. 10 cm
b. 30 cm

c. 40 cm
Answers:
a. wavelength of lowest resonance = 40 cm (10 x 4)
f = 35000/40 = 875
R1 = 875 (R1 = frequency of resonance number 1)
R2 = 2625
R3 = 4375
b. wavelength of lowest resonance = 120 cm (30 x 4)
f = 35000/120 = 291.7
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R1 = 291.7
R2 = 875.0
R3 = 1458.3
c. wavelength of lowest resonance = 160 cm (40 x 4)
f = 35000/160 = 218.75
R1 = 218.75
R2 = 656.25
R3 = 1093.75
10. Show what the frequency-response curves look like for the tubes in the problem above.
11. A complex periodic signal has a fundamental period of 4 msec. What is the fundamental frequency of the signal? At
what frequencies would we expect to find energy?
12. How are the terms octave and harmonic different?
13. Give examples of the following kinds of graphs, being sure to label both axes:
a. amplitude spectrum
b. phase spectrum
c. frequency-response curve
d. time-domain representation
14. Give a brief explanation of the basic idea behind Fourier analysis. What is the input to Fourier analysis and what kind
of output(s) does it produce?
15. Draw and label frequency-response curves for low-pass, high-pass, and band-pass filters.
16. What parameters control the frequency of vibration of a spring and mass system?
17. Draw the time domain representation of one cycle of a sinusoid as variations in instantaneous air pressure over time
and one cycle of that same sinusoid as variations in instantaneous velocity over time.
18. How, if at all, are the terms resonant frequency and harmonic different?
19. How, if at all, are the terms resonant frequency and formant different?
20. A harmonic is a peak in: (a) a frequency response curve, (b) an amplitude spectrum, or (c) either a frequency response

curve or an amplitude spectrum.
21. A resonance is a peak in: (a) a frequency response curve, (b) an amplitude spectrum, or (c) either a frequency response
curve or an amplitude spectrum.
22. A formant is a peak in: (a) a frequency response curve, (b) an amplitude spectrum, or (c) either a frequency response
curve or an amplitude spectrum.
23. A frequency response curve describes a _____.
24. An amplitude spectrum describes a _____.

Frequency Response Problems

Answers to Frequency Response Problems

Decibel Study Questions
1. What reference is used for the dB IL scale?
2. What reference is used for the dB SPL scale?
3. What reference is used for the dB HL scale?
4. What reference is used for the dB SL scale?
5. A listener barely detects a 125 Hz pure tone at 55 dB SPL. Does this listener have a hearing loss at 125 Hz, and if
so, what is the size of the hearing loss?
6. A listener barely detects a 1,000 Hz pure tone at 55 dB SPL. Does this listener have a hearing loss at 1,000 Hz,
and if so, what is the size of the hearing loss?
7. A listener barely detects a 125 Hz pure tone at 55 dB HL. Does this listener have a hearing loss at 125 Hz, and if
so, what is the size of the hearing loss?
8. A listener barely detects a 1,000 Hz pure tone at 55 dB HL. Does this listener have a hearing loss at 1,000 Hz,
and if so, what is the size of the hearing loss?
9. 60 dB SPL at 1,000 Hz means _____ more intense than _____.
10. 60 dB IL at 1,000 Hz means _____ more intense than _____.
11. 60 dB HL at 1,000 Hz means _____ more intense than _____.
12. The reference that is used for the dB SPL scale is:
a. a number
b. a sentence

13. If the answer to the question above is a number, give the number; if it's a sentence, give the sentence.
14. The reference that is used for the dB HL scale is:
a. a number
b. a sentence
15. If the answer to the question above is a number, give the number; if it's a sentence, give the sentence.
16. A specific individual has a 70 dB hearing loss in the left ear at 1,000 Hz. A 90 dB HL, 1,000 Hz tone that is
presented to this listener's left ear would measure _____ dB SL.
17. A sound measures 42 dB IL. On the dB SPL scale, that same sound will measure:
a. 84 dB SPL because with the dB SPL formula we are now are multiplying the ratio by 20 instead of 10.
b. 42 dB SPL because the two versions of the formula are equivalent
18. A sound measures 60 dB IL. (a) The measured intensity (I M) must therefore be _____ times
greater than the reference intensity (I R). (b) What would the pressure ratio (E M/E R) be for this same sound? (c)
Do the arithmetic to show what this sound would measure in dB SPL.

19. A sound measures 40 dB IL. (a) The measured intensity (I M) must therefore be _____ times
greater than the reference intensity (I R). (b) What would the pressure ratio (E M/E R) be for this same sound? (c)
Do the arithmetic to show what this sound would measure in dB SPL.
20. On the graph below, put a mark at: (a) 3,000 Hz, 20 dB SPL, and (b) 3,000 Hz, 20 dB HL (the
grid lines on the y axis are spaced at 2 dB intervals).

Frequency in Hz	Threshold in dB SPL
125	45.0
250	25.5
500	11.5
750	8.0
1000	7.0
1500	6.5
2000	9.0
3000	10.0
4000	9.5
6000	15.5

8000 13.0
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Answers to Decibel Study Questions
1. 10 -12 watts/m2
2. 20 µPa (or, equivalently, 0.0002 dynes/cm2)
3. The threshold of audibility for an average, normal-hearing listener at a particular signal frequency.
4. 3. The threshold of audibility for a particular listener at a particular signal frequency.
5. Consulting the attached figure and table showing the audibility curve for average, normal-hearing listeners, we
find that the threshold of audibility at 125 Hz is 45 dB SPL. A listener who barely detected a 125 Hz tone at 55
dB SPL would therefore have hearing loss of 55-45=10 dB; that is, the hearing sensitivity of this listener would be
10 dB worse than normal.
6. Consulting the attached figure and table showing the audibility curve for average, normal-hearing listeners, we
find that the threshold of audibility at 1,000 Hz is 7 dB SPL. A listener who barely detected a 1,000 Hz tone at
55 dB SPL would therefore have a hearing loss of 55-7=48 dB; that is, the hearing sensitivity of this listener
would be 48 dB worse than normal.
7. The reference for dB HL is the audibility threshold, so this listener would have a 55 dB hearing loss at 125 Hz.
There is no need to consult the table.
8. The reference for dB HL is the audibility threshold, so this listener would have a 55 dB hearing loss at 1,000 Hz.
There is no need to consult the table.
9. 6 factors of 10 (i.e., 1,000,000 times) more intense than 20 µPa)
10. 6 factors of 10 (i.e., 1,000,000 times) more intense than 10 -12 watts/m2
11. 6 factors of 10 (i.e., 1,000,000 times) more intense than a 1,000 Hz tone that is barely audible to an average,
normal-hearing listener.
12. a number
13. 20 µPa
14. a sentence
15. The threshold of audibility for an average, normal-hearing listener at a particular signal frequency.
16. 20 dB SL. The reference for the dB SL (SL=sensation level) is the threshold of audibility for a specific listener. So,

what we want to know here very simply is where this 90 dB HL tone is in relation to this particular listener's
threshold. This listener has a 70 dB hearing loss at this frequency, so the 90 dBHL tone, which would be 90 dB
above a normal-hearing listener's threshold, is only 20 dB above this particular listener's threshold.
17. 42 dB SPL: The pressure version of the formula was derived from the intensity version through algebraic
manipulations, so they have to be equivalent to one another. The next problem was designed to illustrate how
this can be the case.
18. (a) 1,000,000 times (6 factors of 10) more intense than I R. (b) If the intensity ratio is 1,000,000, the pressure
ratio has to be the square root of 1,000,000, which is 1,000. (c) dB SPL = 20 log 1,000 = 20 . 3 = 60 dB SPL. This is
exactly what we got for the same sound measured in dB IL. It will always be the same. If a sound measures 60
dB IL, that same sound will measure 60 dB SPL.
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19. (a) 10,000 times (4 factors of 10) more intense than I R. (b) If the intensity ratio is 10,000, the pressure ratio has
to be the square root of 10,000, which is 100. (c) dB SPL = 20 log 100 = 20 . 2 = 40 dB SPL. This is exactly what
we got for the same sound measured in dB IL. It will always be the same. If a sound measures 40 dB IL, that same
sound will measure 40 dB SPL.
20. See below. The lower of the two marks is 20 dB (2 factors of 10) above the constant reference line of 20 μPa.
The higher of the two marks is 20 dB (also 2 factors of 10) above the curvey line, which is the threshold of
audibility for the average normal-hearing listener.
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A Tutorial on Digital Sound Synthesis Techniques
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A Tutorial on Digital Sound Synthesis Techniques

Introduction

Progress in electronics and computer technology has led to an ever-increasing utilization of digital techniques for musical sound production. Some of these are the digital equivalents of techniques employed in analog synthesizers and in other fields of electrical engineering. Other techniques have been specifically developed for digital music devices and are peculiar to these.

This paper introduces the fundamentals of the main digital synthesis techniques. Mathematical developments have been restricted in the exposition and can be found in the papers listed in the references. To simplify the discussion, whenever possible, the techniques are presented with reference to continuous signals.

Sound synthesis is a procedure used to produce a sound without the help of acoustic instruments. In

digital synthesis, a sound is represented by a sequence of numbers (samples). Hence, a digital synthesis technique consists of a computing procedure or mathematical formula, which computes each sample value.

Normally, the synthesis formula depends on some values, that is, parameters. Frequency and amplitude are examples of such parameters. Parameters can be constant or slowly time variant during the sound. Time-variant parameters are also called control functions.

Synthesis techniques can be classified as (1) generation techniques (Fig. la), which directly produce the signal from given data, and (2) transformation techniques (Fig. Ib), which can be divided into two stages, the generation of one or more simple signals and their modification. Often, more or less elaborate combinations of these techniques are employed.

Fixed-Waveform Synthesis

In many musical sounds, pitch is a characteristic to which we are quite sensitive. In examining the temporal shape of pitched sounds, we see a periodic repetition of the waveform without great variations. The simplest synthesis method attempts to reproduce this characteristic, generating a periodic signal through continuous repetition of the waveform. This method is called fixed-waveform synthesis.

The technique is carried out by a module called an oscillator (Fig. 2), which repeats the waveform with a specified amplitude and frequency. In certain cases, the waveform is characteristic of the oscillator and cannot be changed. But often it can be chosen in a predetermined set of options or given explicitly when required.

Usually, in digital synthesis the waveform value at a particular instant is not computed anew for each sample. Rather, a table, containing the period values computed in equally spaced points, is built beforehand. Obviously, the more numerous the

points in the table, the better the approximation will be. To produce a sample, the oscillator requires the waveform value at that precise instant. It cyclically searches the table to get the point nearest to the required one. Sometimes a finer precision is achieved by interpolation between two adjacent points.

The distance in the table between two samples read at subsequent instants is called the sampling_increment. The sampling _increment is proportional to the frequency f of the generated signal according to the following formula (Mathews 1969):

N
samplingincrement = SRf,

where N is the table length and SR the sampling rate.

In the oscillator, the frequency is usually speci-

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Fig. 1. Classification of synthesis techniques. Generation techniques (a) and transformation techniques (b).

Fig. 2. Fixed-waveform synthesis oscillator.

(a) Parameters
'[M SoundGeneration S
signal
Complex

(b) Parameters sound
I signal
Generation Transformation
Simple
signals
A(t) f(t)

s(t)
fled as a sampling- increment and the algorithm that realizes it is as follows:

signal [t] := amplitude * table [phase], (Relation 1)

and

phase := mod(n, phase + samplingincrement), (Relation 2)

where

Table contains one period of the waveform;
Phase is the theoretical position in the table of the sample to be extracted at the instant; and
Amplitude is the signal amplitude.

Relation 2 computes the phase value in the subsequent instant, approximating the frequency integration by a summation. The modulus operation keeps the phase inside the table length n.

It is noteworthy that the signal generated in this way is an approximation of the desired one (Mailliard 1976). The approximation depends on the table length, the interpolation method, and the signal frequency. For a sufficiently long table, it is fully satisfactory.

The results of fixed-waveform synthesis are of poor musical quality, as the sound does not present any variation along its duration. This technique can be changed by allowing the amplitude to vary in time. In real sounds, the amplitude is rarely constant: it starts from zero, reaches a maximum after a certain time (attack), remains nearly constant (steady state) and, after a certain evolution, it returns to zero (decay). This sequence of amplitude behavior is called the envelope. Thus, when the amplitude varies according to a control function, we have fixed-waveform synthesis with an amplitude envelope.

The envelope can be generated in many ways. In software-based synthesis, the most frequent method uses an oscillator module, seen previously, using a very low frequency equal to the inverse of

the duration. In this case, it performs a single cycle and its waveform corresponds to the amplitude envelope.

By carefully analyzing natural periodic sounds, it has been shown that even the most stable ones contain small frequency fluctuations. These improve the sound quality and avoid unpleasant beatings when more sounds are present at the same time.

The fixed-waveform technique can also be modified so that the oscillator frequency can slowly vary around a value. This enables the production of a tremolo and, with wider variations, of a glissando or melodies.

The combination of these two variations constitutes fixed-waveform synthesis with time-varying amplitude and frequency. The waveform is fixed, while the amplitude and frequency vary. The partials are exact multiples of the fundamental, and they all behave the same.

Fixed-waveform synthesis is realized rather simply. Hence, it is often employed when good sound quality is not required. The constant waveform gives the sound a mechanical, dull, and unnatural character, which soon annoys the audience. Thus,

in musical applications, fixed-waveform synthesis is not very effective when used alone. It is employed for its simplicity when timbral variety is not required, for example, for real-time synthesis on very limited hardware.

For economy, other methods of generating waveforms that do not use tables or multiplications have been devised. The simplest generates a square or (more generally) a rectangular wave, alternating sequences of positive and negative samples of the same value. The frequencies that can be obtained are submultiples of the sampling rate.

A sawtooth signal can also be generated by an ac-

cumulator to which a constant value is continuously added. The output increases linearly until it overflows and starts from the beginning. The signal frequency is proportional to the constant value. This method is used to produce linearly variable control signals. Every time the additive constant changes, the slope changes. Hence, functions composed of straight segments, such as envelopes, can be obtained.

This technique has been generalized recently by Mitsuhashi (1982a). A polynomial of degree N can be generated by putting N accumulators in cascade. The accumulators are initialized by the value of the forward differences, in decreasing order, of the polynomial to be generated (Cerruti and Rodeghiero 1983). The waveforms obtained exhibit great variety and, in certain conditions, they are periodic.

Granular Synthesis

The technique of fixed-waveform synthesis produces rather static sounds in time. Yet a fundamental characteristic of musical sound is its timbral evolution in time. A sound can be thought of as a sequence of elementary sounds of constant duration, analogous to a film, in which a moving image is produced by a sequence of images.

In computer music, the elementary sounds are called grains, and the technique of exploiting this facility is granular synthesis (Roads 1978). The grains can be produced by a simple oscillator or by other methods. The duration of each grain is very short, on the order of 5-20 msec.

There are two ways to implement granular synthesis. The first is to organize the grains into frames, like the frames of a film. At each frame, the parameters of all the grains are updated. This is the approach sketched by Xenakis (1971). The second way involves scattering the grains within a mask, which bounds a particular frequency/amplitude/time region. The density of the grains may vary within the mask. This is the method imple-

mented by Roads (1978).

A problem with granular synthesis is the large amount of parameter data to be specified. In some other types of synthesis (additive and subtractive, to be discussed shortly), these data can be obtained by analyzing natural sounds. However, no analysis system for granular synthesis has been developed. Another possibility is to obtain the parameter data from an interactive composition system, which allows the composer to work with high-level musical concepts while automatically generating the thousands of grain parameters needed.

Additive Synthesis

In additive synthesis, complex sounds are produced by the superimposition of elementary sounds. In certain conditions, the constituent sounds fuse together and the result is perceived as a unique sound. This procedure is used in some traditional instruments, too. In an organ, the pipes generally produce relatively simple sounds; to obtain a richer spectrum in some registers, notes are created by using more pipes sounding at different pitches at the same time. The piano uses a different procedure. Many notes are obtained by the simultaneous percussion of two or three strings, each oscillating at a slightly different frequency. This improves the sound intensity and enriches it with beatings.

In order to choose the elementary sounds of additive synthesis, we first note that the Fourier analysis model enables us to analyze sounds in a way similar to the human ear and so to extract parameters that are perceptually significant. When we analyze a real, almost-periodic sound, we immediately notice that each partial amplitude is not proportionally constant, but that it varies in time

Fig. 3. Additive synthesis.

AI(t) f,(t) A2(t) f2(t) AM(t) fM(t)

$s(t)$

according to different laws. In the attack portion of a note, some partials, which in the steady state are negligible, are often significant.

Any almost-periodic sound can be approximated as a sum of sinusoids. Each sinusoid's frequency is nearly multiple that of the fundamental, and each sinusoid evolves in time. For higher precision, the frequency of each component can be considered as slowly varying. Thus, additive synthesis consists of the addition of some sinusoidal oscillators, whose amplitude, and at times frequency, is time varying (Fig. 3).

The additive-synthesis technique also provides good reproduction of nonperiodic sounds, presenting in the spectrum the energy concentrated in some spectral lines. For example, Risset (1969) imitated a bell sound by summing sinusoidal components of harmonically unrelated frequencies, some of which were beating. In Risset's example, the exponential envelope was longer for the lower partials.

Additive synthesis provides great generality. But a problem arises because of the large amount of data to be specified for each note. Two control functions for each component have to be specified, and normally they are different for each sound, depending on its duration, intensity, and frequency. The possibility of data reduction has been investigated. At Stanford University, a first result has been obtained by representing the control functions of the amplitude and the frequency of each component by line segments, without affecting "naturalness" of the sound (Grey and Moorer 1977).

The next step has been to investigate the relations between these functions (Risset and Mathews 1969; Beauchamp 1975) or their relation to others of more general character (Charbonneau 1981). Additive synthesis is most practically used either in synthesis based on analysis (analysis/synthesis), often transforming the extracted parameters, or

when a sound of a precise and well-determined characteristic is required, as in psychoacoustic experiments. In any case, in order to familiarize musicians with sound characteristics and frequency representations, the technique is also useful from a pedagogical point of view.

Additive synthesis can be generalized by using waveform components of other shapes besides sinusoids. To allow the reproduction of any sound, these waveforms have to satisfy specific mathematical properties. Walsh functions are an example of this kind of function; they are used for their simple hardware realization (Rozenberg 1979).

VOSIM

In the synthesis techniques already discussed, oscillators that periodically reproduce a given waveform are employed. Other synthesis techniques, instead of continuously repeating a given waveform, calculate it anew each period, with minor variations. The control of this calculation process allows continuous spectral variations. A common method of this type is the voice simulation (VOSIM) technique. A VOSIM oscillator has been devised in a project at the Institute of Sonology in Utrecht (Kaegi 1973, 1974; Kaegi and Tempelaars 1978).

The VOSIM waveform (Fig. 4) consists of a sequence of N pulses of shape sin2, of the same duration T, and of decreasing amplitude. The sequence is followed by a pause M. Each pulse's amplitude is smaller than the preceding one, by a constant factor b.

The VOSIM spectrum (Fig. 5a) is described as the product of two terms (Tempelaars 1976; De Poli and De Poli 1979). The first term S, (Fig. 5b) depends only on the pulse shape and limits the signal bandwidth to 2F (being F = 1/T). The second term S2 (Fig. 5c) depends on the relationship between the individual pulse amplitudes. S2 is periodic in the frequency domain with a period F, and it is sym-

Fig. 4. VOSIM oscillator: T is the duration of single pulse, M the rest between two sequences of pulses.

Fig. 5. Spectral envelope of a VOSIM oscillator (N = 5, b = 0, 8) (a). The envelope is the product of the terms S, (b) and S2 (c).

1.5
1
0.5
0 5 M- 10 15
T

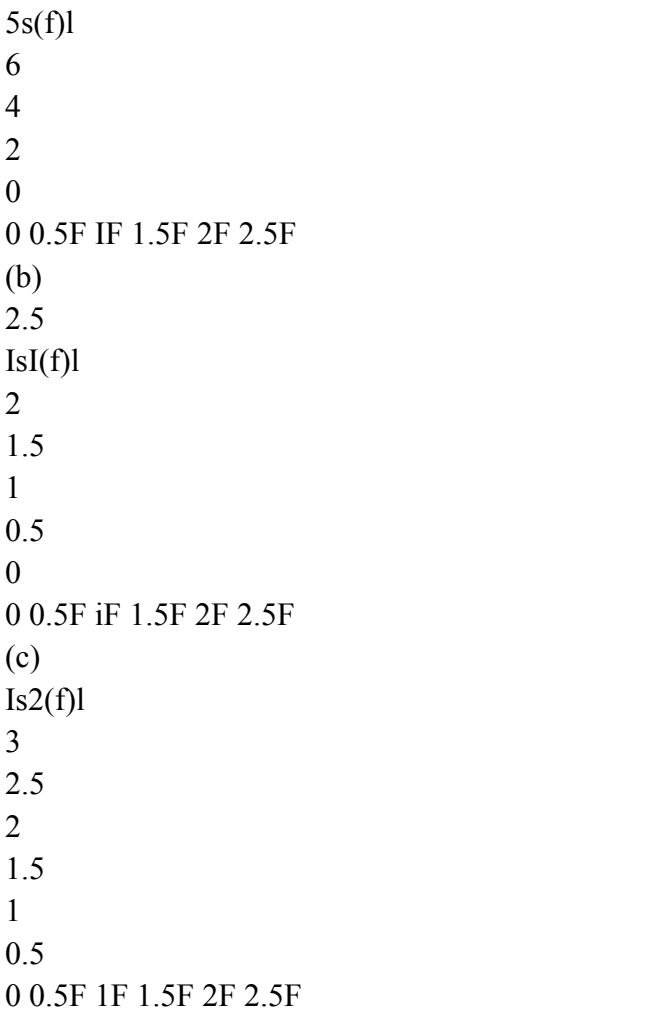
metric with respect to F/2. When b 1, its amplitude will be greater around the extremes of the period 0 and F. When b - -1, its amplitude will be greater in the central position around F/2. Thus, a characteristic formant in F or F/2 will result. The number of pulses N produces N oscillations in the S2 term between 0 and F, with strong signals for b near +F.

This constitutes the spectral envelope of the repeated waveform. Taking a as the ratio between the signal period and a single pulse duration, the number of the harmonic corresponding to the formant is a if b is positive, and a/2 if b is negative. Thus, by varying a, the formant shifts, and the relative amplitude of all the harmonics vary continuously but not homogeneously, following the spectral envelope. The signal and the formant frequencies can be separately controlled.

More kinds of sounds can be obtained by modulating (sinusoidally or randomly) the value of the time interval M between two consecutive pulse sequences. This means that a varies independently from T. In this case, the formant frequency remains constant while the harmonic amplitudes vary. Then

the ear can easily perceive the spectral envelope and fuse the components together. This property makes the VOSIM oscillator effective in musical applications.

If a variation is strong, practically aperiodic sounds or colored noises are obtained. Adding several VOSIM oscillators allows one to control the position of the formants. This results in an additive



synthesis of already complex sounds rather than of sinuosidal components. Instead of the frequency of partials, the position of the formants is controlled. This is a more relevant parameter, from an acoustic

standpoint.

The formant-wave-function synthesis of Rodet (1980) is analogous to VOSIM, but it allows overlapping of single waveforms. This provides better control and generally richer sounds. Mitsuhashi (1982a) and Bass and Goeddel (1981) generalized the VOSIM model by including the case of pulses of any amplitude and using different elementary waveforms.

Synthesis by Random Signals

Up to now, we have considered signals whose behavior at any instant is supposed to be perfectly knowable. These signals are called deterministic signals. Besides these signals, random signals, of unknown or only partly known behavior, may be considered. For random signals, only some general characteristics, called statistical properties, are known or are of interest. The statistical properties are characteristic of an entire signal class rather than of a single signal. A set of random signals is represented by a random process. Particular numerical procedures simulate random processes, producing sequences of random (or more precisely, pseudorandom) numbers. The linear congruential method is commonly used to produce uniformly distributed numbers. From a starting value X0, a sequence of random integers X0, X1, . . . , XK... is generated according to the relation

XK+ = (a XK + C)modm,

where m is the modulus and the maximum sequence period, and a and c are two specific integer constants.

The modulus operation can be avoided by choosing m as the maximum number representable in the computer, that is, m = 2b, where b is the word length (bit number in a binary computer). So the numbers are automatically truncated. The choice of X0, a, and c greatly affects the statistical characteristics of the generated sequence, and its acceptability has to be accurately verified by statisti

tests. A general discussion of various distribut
and the methods used to generate them can be
in Lorrain's paper (1980).
Random sequences can be used both as sign
(i.e., to produce white or colored noise used as
put to a filter) and as control functions to pro
a variety in the synthesis parameters most per
tible by the listener.
In the analysis of natural sounds, some chara
teristics vary in an unpredictable way; their m
statistical properties are perceptibly more sign
cant than their exact behavior. Hence, the add
of a random component to the deterministic f
tions controlling the synthesis parameters is o
desirable.
In general, a combination of random processe
is used because the temporal organization of th
musical parameters often has a hierarchical asp
It cannot be well described by a single random
process, but rather by a combination of rando
cesses evolving at different rates.

Linear Transformations

Let us now examine techniques for signal modifica-
tion. A transformation is a set of rules and proce-
dures transforming a signal called input to another
signal called output. A transformation is linear if
the superimposition principle is valid, that is, if the
effect of the transformation caused by a two-signal
addition is equal to the addition of the individual
signal transformations applied separately. In partic-
ular, in a linear transformation a signal can be mul-
tiplied by a constant but not by another signal.
Digital filters are linear transformations that can
be described by the following difference equation:

N M
Sa, yK=0 i1=0
where aK and b-
and y(i) are the i
signal. The value
ear combination

with the precedi
DePoli 13

Fig. 6. Finite-impulse-response (FIR) filter with two zeros described by the equation y(n) = x(n) + a,x(n - 1) + a2x(n - 2)
(a). Infinite-impulse-response (IIR) filter with two poles described by the equation y(n) = x(n) + P1y(n - 1) + 32y(n - 2)
(b).
x(n) (a)y(n)
Z-1
x(n - 2)
----'--
x(n) (b)y(n)
Z-1
- ~ y(n- 2)
the input is sinusoidal, the steady-state output is sinusoidal with the same frequency. The amplitude and phase of the frequencies are determined by the system. That is why this transformation is called a filter.

Subtractive Synthesis

Sound produced by filtering a complex waveform is called, sometimes inappropriately, subtractive synthesis. First, a periodic or aleatoric signal rich in harmonics is generated by the previously examined techniques or others. This signal must contain energy in all frequencies required in the output sound. Second, one or more filters are used to alter selectively the specific frequency components. The undesired components are attenuated (subtracted) and others are eventually amplified. When the filter coefficients change, the frequency response changes, too. Thus, it is possible to vary characteristics of

the output sound.

In modular diagrams, filters are usually represented by rectangles and the difference equation or the transfer function is given as a label near the rectangle. Two examples of simple digital filters, showing their internal structure, are shown in Fig. 6. The first filter (Fig. 6a) has a finite-impulse response (FIR). This structure is useful to produce transmission zeros: that is, it can nullify some frequencies that depend on a1, a2 values and on the sampling rate. The second filter (Fig. 6b) is recursive, or has an infinite-impulse response (IIR). Feedback in the structure amplifies certain frequencies, that is, produces transmission poles. When used as bandpass filter, in general terms, the coefficient P, controls the center frequency and the coefficient 32 the bandwidth.

One of the most attractive aspects of digital filtering is that it is analogous to the functioning of many acoustic musical instruments. Indeed, instrument physics can be used as a model for synthesis. For example, in the brasses and woodwind instruments, the lips or vibrating reed generate a periodic signal rich in harmonics. The various cavities and the shape of the instrument act as resonators, enhancing some spectral components and attenuating others. In the human voice, the excitation signals are periodic pulses of the glottis (in the case of voiced sounds) or white noise (in the case of unvoiced sounds-for example, the consonants s and z). The throat, the mouth, and the nose are the filtering cavities, and their dimensions vary in time. Their great variability makes the human voice the most rich and interesting musical instrument.

Today, subtractive synthesis is the standard means of speech synthesis. An analysis procedure, called linear predictive coding (LPC), allows us to obtain

Fig. 7. Elementary filters
used in reverberators.
Comb filters (a). All-pass
filter (b).

the pitch and the coefficients of a recursive (poles only) filter (see Cann's [1979-1980] tutorial and Moorer's paper [1979a]). These data can be utilized to synthesize the sound directly or following modification. For example, speech can be accelerated or slowed down, and pitch can be varied. An instrument or orchestral sound can be used as input to the filter, producing the effect of a "talking orchestra." Interesting possibilities for musique concrete sound processing arise. Not only simple filtering of sounds is possible, but the modification of their most intrinsic characteristics is also made possible by varying the parameters of the deduced sound-production model.

Generally, LPC is relatively difficult to use. Intuitively, the filter characteristics depend on the position of the zeros and the poles in the transfer function. These characteristics are affected in a complex and nonintuitive way by the filter coefficients. In some simple cases, approximate formulas give the coefficients as functions of significant parameters, that is, center frequency and bandwidth, or cutoff frequency and slope. The filters can be used in series or in parallel. In the most complex cases, a precise analysis is obtained by using specific programs for digital filter design and analysis. Such digital filters can be very stable and precise, but only at the cost of a large amount of calculation. Simple linear digital networks can also be used as oscillators (Tempelaars 1982) by applying a pulse sequence to the input and choosing an impulse response equal to the signal function to be generated.

Reverberation

One application of digital filters is sound reverberation. An acoustic environment can be simulated by

distributing sound among different loudspeakers and by adjusting the ratio between direct and reverberated sound (Chowning 1971). Most of the studio reverberators sold today use digital technology.

The two elementary filters used in reverberation are shown in Fig. 7. The first filter is called a comb filter; in it, the signal is delayed a certain number of samples, attenuated, and added to the input. An exponentially decaying, repeated echo is so obtained. The frequency response is characterized by equispaced peaks-hence this filter's name. The peaks' amplitude increases as G approaches 1.

(a)
+)(Delay
(b)
- G

The second filter is called an all-pass filter, since the frequency response is flat and there is only a phase shift. The input signal is attenuated and subtracted from the delayed signal so that the feedback effect is compensated and the echoes are maintained. The all-pass property is valid only in the steady state with stationary sounds, not in transient states. Thus, it has a well-defined sound quality that a skilled listener can easily distinguish.

Reverberators are built combining some of these filters (Moorer 1979b). Distinguishable signal repetitions should not occur in them, since the reverberated result should consist of a diffused sound. The delay time of each elementary filter has to be chosen very carefully. Sometimes a nonrecursive echo generator is added to produce the first aperiodic echoes, which are the main perceptual determinants of the characteristics of the room.

Nonlinear Techniques

In addition to linear transformations, which are used in other fields and have a rather developed theory, nonlinear transformations are used more and

Fig. 8. Waveshaping.

more commonly in musical applications. They derive mainly from electrical communication theory, and they have proved to be promising and effective. One use of nonlinear synthesis is in the large amount of computer music generated by frequency modulation (FM) synthesis (Chowning 1973).

In the classic case, nonlinear techniques use simple sinusoids as input signals. The output is composed of many sinusoids, whose frequency and amplitude depend mostly on the input ones.

Two main types of nonlinear techniques can be distinguished, waveshaping and modulation. In waveshaping, one input is shaped by a function depending only on the input value in that instant. In modulation (with two or more inputs), a simple parameter of one signal, called the carrier, is varied according to the behavior of another signal, called the modulator. In electrical communications (e.g., radio) the spectra of the signals are clearly distinguished and therefore easily separable. The originality in computer music application is the utilization of signals in the same frequency range. Thus, the two signals interact in a complex way, and simple input variation affects all the resultant components.

Often, the input amplitudes are varied by multiplying them by a constant or time-dependent parameter I, called the modulation index. Thus, acting only on one parameter, the sound characteristics are substantially varied. Dynamic and variable spectra are easily obtainable. In additive synthesis, similar variations require a much larger amount of data.

Waveshaping

A linear filter can change the amplitude and phase of a sinusoid, but not its waveform, whereas the aim of waveshaping is to change the waveform. The distortion of a signal heard from a nonlinear ampli-

fier is common. The output from a nonlinear amplifier of a sinusoidal signal is a signal with the same period, but with a different waveform. The various harmonics are present, and their amplitude depends on the input and on the distortion. In stereo systems, these distortions are usually avoided,

x(t)

y(t)y~t)

while waveshaping (Arfib 1979; Le Brun 1979; Roads 1979) exploits them to generate periodic sounds, rich in harmonics, from a simple sinusoid. The function F(x), describing distortion, is called the shaping function, and it associates with each input value the corresponding output value independent of time. If the input is x(t) = cos(27r ft), the output is

s(t) = F(x(t)) = F(cos[2rr ft]).

In analog synthesis, it is difficult to have an amplifier with a precise and variable distortion characteristic. In digital synthesis, this technique is extremely easy to implement (Fig. 8). As in the case of the oscillator, the shaping function can be previously computed and stored in a table. All that is necessary is to look up the proper value from the table.

Generally, if F(x) = F,(x) + F2(x), the distortion produced by F is equal to the sum of those produced by F, and F2 separately. Usually, the shaping produces infinite harmonics. But when a polynomial of degree N is chosen as shaping function, only the first N harmonics are present. Thus, foldover is easily avoided. Arfib and Le Brun deal extensively with the mathematical relations among the coefficients di of the shaping polynomial and the amplitudes hi of harmonics generated when the amplitude I of the cosinusoidal input varies.

The shaping function, producing the jth harmonic, is the Chebychev polynomial 7T(x) of degree i (Fig. 9). Thus, to obtain the various harmonics of

Fig. 9. Chebychev poly-
nomial of degree K used as
shaping function produces
only the Kth harmonic. In
the figure, K = 3.
T3(cos[ot]) = cos(3ot)
T3(x) = 4X3 - 3x
X = cos(wt) wt
01 1
-rr/6 -
2I/37T/3_ os(ut)
73/2
27r/3
1lT/6
77r/6
137T/6
ot
amplitude h,, it is sufficient to add the correspon-
dent Chebychev polynomials, each multiplied
by hi:
N N
F(x) = h(x) = d;x.=0 i=O0
From these relations, it follow
monics are comp
even polynomial
harmonics. In th
only the odd ha
cient of x7 affect
enth harmonics
harmonic of ord
odd) coefficients
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example, the seventh harmonic is affected by the
odd coefficients from the seventh up to the degree
of the polynomial.
When the input amplitude I varies, the distortion

and the output spectrum vary. This is similar to an expansion or contraction of the function, since greater or smaller range of the function is employed. From a mathematical point of view, the amplitude variation corresponds to the multiplication of each polynomial coefficient d, by II. The amplitudes of the even or odd harmonics depend on I according to the even (or odd) polynomials, which contain the terms from the harmonic order up to the polynomial degree.

If the spectrum is rather smooth, the number of significant harmonics increases with the index. Thus, a typical characteristic of real instruments is reproduced, in that amplitude and spectrum are correlated. The amplitude and loudness of the output vary with the input amplitude. In simple cases, this effect can be compensated for by multiplying the output by a suitable normalization function. But in musical applications, the amplitude of the signal is rarely constant, and it is multiplied by an envelope. Normalization can be avoided by combining it with the amplitude envelope in experimental or intuitive ways after considering the normalization function.

It is also advisable to choose the even (or odd) polynomial coefficients with alternating signs, that is, according to the following model: + + - - + + - -. It is also advisable that the hi amplitude not decrease abruptly, sharply limiting the band. Otherwise, a spectrum would result that varied very irregularly with I.

Dynamic spectral behavior cannot be easily anticipated from the coefficients or from the static spectrum. Moreover, the same (absolute-value) spectrum can be produced by many polynomials with different dynamic behaviors (Forin 1982). With waveshaping, listening and graphic considerations have more relevance than purely mathematical formulations.

Another dynamic variation of waveshaping that is easy to implement occurs when a constant is

added to the input; the shaping function shifts horizontally. Even in this case, the spectrum varies. The signal is periodic, with the same number of harmonics. But in this case, the harmonic behavior depends on both the even and the odd coefficients. Generalizations of waveshaping technique are possible. Reinhard (1981) studied the relations that produce the partials generated by the polynomial distortion of two cosine waves of frequency f, and f2. All the components of frequency 1K f, ? Jf2 with IK + j|l N, where N is the polynomial degree, are present.

Shaping functions that are not polynomial can be used if the spectra produced by them are almost band limited. Of particular interest is the use of trigonometric and exponential functions (Moorer 1977) and of those where the input also appears in the denominator (Winham and Steiglitz 1970; Moorer 1976; Lehmann and Brown 1976; De Poli 1981).

Due to the wide spectral variation induced by only one parameter (amplitude or shift), waveshaping is particularly convenient in musical applications, especially in combination with multiplicative synthesis. Moreover, it is suitable for modeling the sound production of some acoustic instruments (Beauchamp 1979, 1982). There is a large and not intuitive problem in choosing the coefficients, however, and further research is required.

Multiplicative Synthesis (Ring Modulation)

The simplest nonlinear transformation consists of the multiplication of two signals. In analog synthesizers, it is called ring modulation (RM). Sometimes it is also called amplitude modulation (AM), but the two differ, especially in their realization. With two inputs x,(t) and x2(t), the output is s(t) = Xl(t) . x2(t). Obviously, when the inputs interchange, the result does not vary. The resulting spectrum is obtained from the convolution of the two signals' spectra. Usually, one of the two signals,

called the carrier, is sinusoidal; the result is not too complex and noisy.

When x1 is the sinusoidal carrier of frequency f/, and x2 (modulator) is sinusoidal with frequency f2, from cos(a) , cos(3) = 12lcos(a + 3) + cos(a - /)1,

Fig. 10. Multiplicative synthesis. Spectrum of a periodic signal X2 with four harmonics (a). Resulting spectrum when d2 is multiplied by a sinusoid of frequency f, greater than its bandwidth (f, = 7f2) (b). Resulting spectrum when x2 is multiplied by a sinusoid of frequency inferior to its bandwidth (f, = 26f2) (c). The components deriving from the folding of negative frequencies are shown as dashed lines.

the output consists of two sinusoidal partials of frequency fI + f2 and f, - f2. The phases of the output are also the sum and the difference of the phases of the two inputs. For example, if x, and x2 frequencies are 400 Hz and 100 Hz, the output has two partials of frequency 500 Hz and 300 Hz.

Negative frequencies may occur, for example, when f, = 100 Hz and f, = 400 Hz. This often happens in modulations (foldunder) and can be explained by the trigonometric relation cos(a) = cos(-a), from which cos(27r ft + (4) = cos(7r[-flt - 4). The alteration of the frequency sign only changes the sign of the phase with respect to the cosine. In particular, a cosine signal is unaffected, while a sine wave changes its sign. In the interpretation of the results, only absolute frequency

values have to be considered. Usually, the phase is not significant, as the ear is not terribly sensitive to it. But the phase has to be taken into account while summing the amplitude of components of identical frequencies.

In multiplicative synthesis, usually x2 is periodic with frequency f2. The multiplication causes every harmonic spectral line of frequency K . f2 in the original signal to be replaced by two spectral lines (called sidebands) of frequency f, + K f2 and f, - K f2. The resulting spectrum has components of frequency If +? K f2 , where K is equal to the order of the different harmonics in x2 (Fig. 10). Thus two sidebands, symmetric with respect to the carrier, occur. When f, is less than the greatest frequency in x2, then the negative frequencies fold around zero, as discussed above.

The possibility of shifting the spectrum is very intriguing in musical applications. From simple components, harmonic and inharmonic sounds can be created, and various harmonic relations among the partials can be established. If x2 is a signal with spectrum X2, the signal obtained from its multiplication with a sinusoid of frequency f, has two sidebands symmetric with respect to f, and shaped like X2.

A periodic signal x, can be expanded in Fourier series. Each x, partial will have sidebands of amplitude proportional to its own. If f1 is less than the bandwidth of x,, then the sidebands overlap with

(a) jx2(f)l

(b)

I|S(f)l

ff2

(c)

S(f)I

Iu I li I fl

eventual component superimposition. In this case, the phases have to be taken in account while summing. Dashow (1978, 1980) describes some general-

ization of this technique and employs the generated spectra for particular "harmonizations" of pitches specified by the composer.

Amplitude Modulation

In RM, the carrier does not appear in the spectrum created by the product of a sinusoidal carrier with another signal, except when the modulator has a direct current (dc) component. In carrying out the modulation in AM (Fig. 11), the carrier is present in the output, with an amplitude independent of the sidebands. The formula for AM is as follows:

s(t) = xl(t) . (K + x2(t)).

The result is RM with carrier added. When the carrier is sinusoidal and the modulator is periodic, the spectrum is composed of partials of frequency If1 1 K f,2, with K = 0, 1, . It is useful to distin-

Fig. 11. Amplitude modulation.

K A2 f2 f1

X2(t)

x1(t)

s(t)

guish between the two modulations because they have different realization schemes.

Spectra of Type If , K f2

The following considerations are valid for all spectra whose components are of type If, + K f2h, with K = 0, 1, ... The spectrum is characterized by the ratio f,/f2. (This is often referred to as the carrier-to-modulator [c:m] ratio.) When this ratio is rational, it can be expressed as an irreducible fraction fI/f2 = NI/N2, with N, and N2 as integers that are prime between themselves. In this case, the resulting sound is harmonic, since the various components are a multiple of a fundamental according to integer factors. The fundamental frequency is

fo = - /

N1 N2 '
and the carrier coincides with the NIth harmonic. If N2 = 1, all the harmonics are present and the sideband components coincide. If N2 = 2, only odd harmonics are present and the sidebands superimpose. If N2 = 3, the harmonics that are multiples of 3 are missing. The c: m ratio is also an index of the harmonicity of the spectrum. The sound is mo "harmonious" intuitively when the N /N2 rati simple and formally when the N, N2 productsmaller. The ratios can be grouped in families (Truax All ratios of the type If, K f2l/f2 can produce same components that flf, produces. Only th tial coinciding with the carrier (f,) changes. Fo ample, the ratios 2/3, 5/3, 1/3, 4/3, 7/3 and s all belong to the same family. Only the harmo that are multiples of 3 are missing (see N2 = 3 the carrier is respectively the second, fifth, fir fourth, seventh, and so on harmonic.

The ratio that distinguishes a family is defin normal form when it is - 1/2. In the previous ample, it is 1/3. Each family is characterized b ratio in normal form. Similar spectra can be p duced using ratios from the same family. Diffe spectra are obtained by sounds of different fa When the fl/f2 ratio is irrational, the resulti sound is aperiodic and hence, inharmonic. Of p ticular interest is the case of an f,/f2 ratio app mating a simple value, that is,

fl/f2 = N1/N2 + e.

Here the sound is no longer rigorously periodi The fundamental frequency fo is still f2/N2, the harmonics are shifted from their exact valu by +e / f2. When N2 is equal to 1 or 2, the po and negative components are not superimposed beat with a frequency of 2e / f2. Hence, a smal of the carrier does not change the pitch, even slightly spreads the partials and makes the sou more lively. But the same shift of the modula frequency f, changes the sound's pitch.

Frequency and Phase Modulation

Another type of modulation, suggested by Cho
ing (1973), has become one of the most widely
synthesis techniques. In general, it consists of
modulation and it can be realized both as ph
modulation (4M) or as FM. This technique do
not derive from models of production of phys
sounds, but only from the mathematical prope
ties of a formula. It has some of the advantage

Fig. 12. The number of sig-
nificant sidebands in FM.

waveshaping and RM, and it avoids some of their
drawbacks.

The technique consists of the modulation of the
instantaneous phase or frequency of a sinusoidal
carrier according to the behavior of another signal
(modulator), which is usually sinusoidal. It can be
expressed as follows:

s(t) = sin(2zT f t + I sin[27r fmt]) =

=>K JK (I)sinj27T(f, + Kfj)tI.

The resulting spectrum is of the type If, +? K f,,. All
the spectral considerations discussed previously are
applicable, particularly those regarding negative fre-
quency, foldunder, fclf, ratios, and harmonic andinharmonic sounds.

The amplitude of each Kth side component of
the FM technique is given by the Bessel function of
Kth order computed in I. To plot the spectrum, a
table of Bessel functions has to be referenced to ob-
tain the amplitudes of the carrier and of the side
frequencies in the upper sideband. The odd-order
side frequencies in the lower sideband have signs
opposite to those in the upper one, and the even-
order side frequencies have the same sign. The
negative frequencies, being sine waves, are folded,
changing the sign. When superimposition occurs,
the amplitudes are added algebraically.

When I (called the modulation index) varies, the

amplitude of each component varies as well. Thus, dynamic spectra can be obtained simply by varying this index. Each component varies its amplitude by following the corresponding Bessel function. A Bessel function can be asymptotically approximated by a damped sinuosoid. So when the index varies, some components increase and others decrease, all without sharp variations.

In Eq. (1), the sum includes infinite terms, so theoretically the signal bandwidth is not limited. But, practically, it is limited. In the Bessel function's behavior, only a few low-order functions are significant for small index values. When the index increases, the number and the order of the significant functions increase. For a given index, the side amplitudes oscillate with gradually increasing amplitude and slowly increasing period all the way from

25
20 /
15 /
5
M ' I I
I 5 10 15 20

the origin to a
toward zero. Th
slightly below
Usually, in the
signal, all side f
than /loo of th
ered. The numb

M = I + 2.4 J10.27

(See Fig. 12.) Often, as a rule of thumb, it is roughly considered as

M = I+ 1.

In Eq. (1), the sum can be performed for K fr
-M to +M. For a harmonic sound, that is, w
the ratio fc/fm = NI/N2 is simple, the maxim
der of significant harmonics is N, + M ' N2.
For wide index variations, the sounds produc
are characteristic of the FM technique. A typi

timbre of FM sound is easily recognizable an
well defined. This does not happen for small i
variations or for compound carriers or modul
Frequency modulation synthesis has another p
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Fig. 13. Frequency
modulation.
A(t) fc d(t) fm
s(t)
Fig. 14. Frequency modu-
lation with N carriers
modulated by the same
oscillator.
a, f/ a2 f2 aN fN d f,
+ + +
s(t)
erty that is very important in musical applications:
the maximum amplitude and the signal power do
not vary with the index I. Unlike the situation in
waveshaping, normalization of the output is not
necessary.
Let us now examine the difference between OM
and FM. Phase modulation is defined as follows:
s(t) = sin (2M ft + 0[t]),
and it corresponds to Eq. (1) if the modulating sig-
nal is 0(t) = I sin(2rr f, t).
Frequency modulation occurs when the instanta-
neous frequency varies around the carrier value ac-
cording to the behavior of the modulating wave. For
a signal s(t) = sin(p[t]), the instantaneous frequency
is fj = (1/27r) (di[t]l/dt). Thus, the instantaneous
frequency of the signal in Eq. (1) is as follows:
fi = fc + I fm , cos(2r f,, t).
The frequency varies around f, with a maximum
deviation d = I f,. Thus, with a modulating wave
I - f, cos(27r f, t), an FM equivalent to OM is ob-
tained. Both phase and frequency modulations are
special cases of angle modulation.

In sound synthesis programs, frequency-driven oscillators are provided. The integration involved in calculating the instantaneous phase is therefore computed automatically. Frequency modulation is normally implemented as in Fig. 13. A change of the phase between the carrier and the modulating wave in Eq. (1) only changes the reciprocal phase of the partials. If components superimpose, their total amplitude changes, and a direct-current component may appear. The next sections examine some useful extensions of the basic algorithm.

Nonsinusoidal Carrier

Here we consider a periodic nonsinusoidal carrier. The result of its modulation is the modulation of each of its harmonics by the same wave. Sidebands of amplitude proportional to each harmonic will be present around the carrier. The result is a spectrum with components of frequency In f, + K . fmi,, with K = 0,... , M and n = 1,..., N, when Nis the number of significant harmonics. The maximum frequency present is N - f, + M . f,. In general, there may be various independent carriers modulated by the same wave (Fig. 14) or by different modulating signals. This is like additive synthesis, only instead of sinusoidal addends, more complex addends are used. For example, harmonic sounds can be generated by controlling the various spectral ranges with a few significant and independent parameters. Sounds of the same "family" are possible.

The frequency of each carrier determines the

Fig. 15. Frequency modulation with two modulators.

Fig. 16. Frequency modulation with N modulators.

A fc dl(t) f, d2(t) f2

s(t)

location of the formant position, the amplitude determines its energy, and the modulation index specifies its bandwidth. Chowning (1981) demonstrated these facilities in synthesis of the singing voice of a soprano.

Compound Modulation

Let us examine the case of a modulation composed of two sinusoids (Fig. 15), each with its own modulation index, applied to a sinusoidal carrier. The formula for two-sine-wave OM (Le Brun 1977) is as follows:

s(t) = sin(121T fct + I, sin[27r f t] + 12 sin[27r f2t])
= K n, JK(I1) Jn(12) . sin(I27T[fc + K f, + n f2ltI).

The same result can be obtained with FM using as modulating signal the following expression:

I,f,/cos(27r f, t) + I2f2cos(2r f2t).

The resulting spectrum is much more complex than in the one-modulator case. All the components of frequency If/ ? K f, n f,2 are present, and their amplitude is JK(I) " Jn(I2).

To interpret the effect, let us consider f1 > f2. If only f, were present, the resulting spectrum would have a certain number of components of amplitude JK(I1) and frequency fc + K f1. When the modulator

A fc d1(t) f, d2(t) f2 dM(t) fM
s(t)

f, is applied, these components become carriers, with sidebands produced by f,. The resulting bandwidth is approximately equal to the sum of the two bandwidths.

If the frequencies have simple ratios, the spectrum is of the type If, -K f,,, where now f, is the greatest common divisor of f, and f,. For example, with f, = 700 Hz, f = 300 Hz, and f2 = 200 Hz, the components are 1700 ? K 1001. Thus, by choosing f, and f2 multiples of f,, sounds belonging to the same family as a simple modulation, but with a more complex spectral structure, can be generated.

In general, if the modulating signal is composed of N sinusoids (Fig. 16), the following relations hold:

s(t) = sin (2rT fct + EI sin[2r fs t]
N
= KS, HKS (Is) sin|2ir(fct + Y Ifs)t1.
Thus, all the components of frequency If , Klf/ -. . ? KNfNI, with amplitudes given by the product
of N Bessel functions, are obtained. A very complex spectrum results. If the relations among the frequencies f, are simple, that is, if the modulating wave is periodic, then the spectrum is of the type Ifc ? K fm, where f,m is the greatest common divisor among the modulating components. Otherwise, the sonorities are definitely inharmonic and particularly noisy for high indexes.

Fig. 17. Nested FM.

Nested or Complex Modulation

Let us examine the case of a sinusoidal modulator that is phase modulated by another sinusoid. The signal is defined as follows:

s(t) = sin(I27r ft + I sin[2ir f, t + 12 sin{2ir f2t}]))
= K JK(Il)sin(I2zr[fc + K f,]t
+ K 12 sin[27r f2t]I)
= 1K J(I,)" Jn(KI2)sin(2r[fc + K,f, + n f2]t).

The result can be interpreted as if each partial produced by the modulator f, were modulated in its turn by f2 with modulation index K I,. Thus, all the partials of frequency If +? K f, n f2j, with approximately 0 - K - I, Os n In , 1 I2, are present. The maximum frequency is fc + I1I(f + 12/2).

The structure of the spectrum is similar to that produced by the two-sinusoid modulation, but with a larger bandwidth. Even where fm is the greatest common divisor between f, and f2, the spectrum is of the type If, + K fn-.

In the equivalent realization by FM (Fig. 17), the spectrum is of the same type, but with slightly different amplitudes. A direct-current component in the resulting modulating wave added to the carrier

is avoided by choosing a sine wave modulated by a cosine wave.

This technique is made more interesting by an algorithm suggested by Justice (1979), which enables an analysis of a sound according to this model, with the frequency and the index behavior of two or more nested modulators being deducible.

Other Two-Input, Nonlinear Transformations

Mitsuhashi (1980) proposed a more complex two-input, nonlinear transformation, in which the instantaneous phase and amplitude of an approximately sinusoidal signal are simultaneously varied. In another paper, Mitsuhashi (1982c) generalized this technique while discussing some criteria in choosing the two-input, nonlinear function and suggesting two examples. The function is time independent, bidimensional, and considered periodic outside the definition field. Thus, it can be implemented with a two-dimensional table, with analogy to an oscillator. This technique appears very inter-

A fc d (t) f/ d2(t) f,

+

s(t)

esting, even if it seems to be difficult to find a simple expression that bounds significant parameters of the resulting spectrum to the input and function characteristics. Another promising modulation technique is linear sweep synthesis, recently suggested by Rozenberg (1982).

Conclusion

As a consequence of progress in digital hardware and software, the initial antithesis between computing efficiency and timbral richness is lessening. Digital sound quality largely depends on the amount of introduced or controlled detail; excessive simplifications lead often to trivial results. It follows that increased computing power can generate more sophisticated results.

A musically interesting sound can be obtained in two ways. The first consists of the utilization of

more complex techniques or of the combination of

many of the techniques described here. Many linear and nonlinear transformations are possible. Most of the parameters do not have to be constant and can be varied by control functions and random signals.

The other synthesis approach consists of the superimposition of many simple sounds produced by basic techniques. The evolution of the individual sounds is not complex, and the richness of the result essentially depends on their combination. In this approach, the parameters of many elementary sounds have to be given. Specific programs are often used to define these parameters.

Sound evolution can be regulated either by control functions in the synthesis or by programs computing the parameters for the synthesis. In any case, many details of the sound have to be accurately controlled. Their coherence both within the sound and in the context of adjacent and simultaneous notes has to be guaranteed. The relations among sounds can be more easily highlighted when they are reflected not only in macroscopic parameter variations but also in internal structure.

The extensive utilization of a single technique reveals its peculiar characteristics. This derives from the finite repertoire of obtainable sounds and, more specifically, from the more easily producible dynamic variations associated with it. Thus, it is wise to use different techniques, the better to exploit their different potential. Moreover, the musician must study and experiment with a technique. This is essential in order to determine all its characteristics and to acquire a feeling for the parameter choices necessary for nontrivial use. In any case, a synthesis technique is simply a tool to produce sound, and sound is not yet music.

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Designing Calm Technology
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Xerox PARC
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Introduction
Bits flowing through the wires of a computer network are ordinarily invisible. But a radically new tool
shows those bits through motion, sound, and even touch. It communicates both light and heavy network

traffic. Its output is so beautifully integrated with human information processing that one does not even
need to be looking at it or near it to take advantage of its peripheral clues. It takes no space on your
existing computer screen, and in fact does not use or contain a computer at all. It uses no software, only
a few dollars in hardware, and can be shared by many people at the same time. It is called the "Dangling
String".
Created by artist Natalie Jeremijenko, the "Dangling String" is an 8 foot piece of plastic spaghetti that
hangs from a small electric motor mounted in the ceiling. The motor is electrically connected to a nearby
Ethernet cable, so that each bit of information that goes past causes a tiny twitch of the motor. A very
busy network causes a madly whirling string with a characteristic noise; a quiet network causes only a
small twitch every few seconds. Placed in an unused corner of a hallway, the long string is visible and
audible from many offices without being obtrusive. It is fun and useful. The Dangling String meets a
key challenge in technology design for the next decade: how to create calm technology.
We have struggled for some time to understand the design of calm technology, and our thoughts are still
incomplete and perhaps even a bit confused. Nonetheless, we believe that calm technology may be the
most important design problem of the twenty-first century, and it is time to begin the dialogue.
The Periphery
Designs that encalm and inform meet two human needs not usually met together. Information
technology is more often the enemy of calm. Pagers, cellphones, newservices, the World-Wide-Web,
email, TV, and radio bombard us frenetically. Can we really look to technology itself for a solution?
But some technology does lead to true calm and comfort. There is no less technology involved in a

comfortable pair of shoes, in a fine writing pen, or in delivering the New York Times on a Sunday

morning, than in a home PC. Why is one often enraging, the others frequently encalming? We believe
the difference is in how they engage our attention. Calm technology engages both the center and the
periphery of our attention, and in fact moves back and forth between the two.
We use "periphery" to name what we are attuned to without attending to explicitly. Ordinarily when
driving our attention is centered on the road, the radio, our passenger, but not the noise of the engine.
But an unusual noise is noticed immediately, showing that we were attuned to the noise in the periphery,
and could come quickly to attend to it.
It should be clear that what we mean by the periphery is anything but on the fringe or unimportant. What
is in the periphery at one moment may in the next moment come to be at the center of our attention and
so be crucial. The same physical form may even have elements in both the center and periphery. The ink
that communicates the central words of a text also, though choice of font and layout, peripherally clues
us into the genre of the text.
A calm technology will move easily from the periphery of our attention, to the center, and back. This is
fundamentally encalming, for two reasons.
First, by placing things in the periphery we are able to attune to many more things than we could if
everything had to be at the center. Things in the periphery are attuned to by the large portion of our
brains devoted to peripheral (sensory) processing. Thus the periphery is informing without overburdening.
Second, by recentering something formerly in the periphery we take control of it. Peripherally we may
become aware that something is not quite right, as when awkward sentences leave a reader tired and
discomforted without knowing why. By moving sentence construction from periphery to center we are
empowered to act, either by finding better literature or accepting the source of the unease and continuing.
Without centering the periphery might be a source of frantic following of fashion; with centering the

periphery is a fundamental enabler of calm through increased awareness and power.
Not all technology need be calm. A calm videogame would get little use; the point is to be excited. But
too much design focuses on the object itself and its surface features without regard for context. We must
learn to design for the periphery so that we can most fully command technology without being dominated by it.
Our notion of technology in the periphery is related to the notion of affordances, due to Gibson by
popularized by Norman. An affordance is a relationship between an object in the world and the intentions, perceptions, and capabilities of a person. The side of a door that only pushes out affords this
action by offering a flat pushplate. The idea of affordance, powerful as it is, tends to describe the surface
of a design. For us the term "affordance" does not reach far enough into the periphery where a design
must be attuned to but not attended to.
Three signs of calm technology
Technologies encalm as they empower our periphery. This happens in two ways. First, as already mentioned, a calming technology may be one that easily moves from center to periphery and back.
Second, a technology may enhance our peripheral reach by bringing more details into the periphery. An
example is a video conference that, by comparison to a telephone conference, enables us to attune to
nuances of body posture and facial expression that would otherwise be inaccessible. This is encalming
when the enhanced peripheral reach increases our knowledge and so our ability to act without increasing
information overload.
The result of calm technology is to put us at home, in a familiar place. When our periphery is

functioning well we are tuned into what is happening around us, and so also to what is going to happen,
and what has just happened. We are connected effortlessly to a myriad of familiar details. This connection to the world around we called "locatedness", and it is the fundamental gift that the periphery
gives us.
Examples of calm technology

To deepen the dialogue we now examine a few designs in terms of their motion between center and
periphery, peripheral reach, and locatedness. Below we consider inner office windows, Internet Multicast, and once again the Dangling String.

inner office windows

We do not know who invented the concept of glass windows from offices out to hallways. But these
inner windows are a beautifully simple design that enhances peripheral reach and locatedness. The hallway window extends our periphery by creating a two-way channel for clues about the environment. Whether it is motion of other people down the hall (its time for a lunch; the big meeting is
starting), or noticing the same person peeking in for the third time while you are on the phone (they
really want to see me; I forgot an appointment), the window connects the person inside to the nearby
world.

Inner windows also connect with those who are outside the office. A light shining out into the hall
means someone is working late; someone picking up their office means this might be a good time for a
casual chat. These small clues become part of the periphery of a calm and comfortable workplace.

Office windows illustrate a fundamental property of motion between center and periphery. Contrast
them with an open office plan in which desks are separated only by low or no partitions. Open offices
force too much to the center. For example, a person hanging out near an open cubicle demands attention
by social conventions of privacy and politeness. There is less opportunity for the subtle clue of peeking
through a window without eavesdropping on a conversation. The individual, not the environment, must
be in charge of moving things from center to periphery and back.

The inner office window is a metaphor for what is most exciting about the Internet, namely the ability to
locate and be located by people passing by on the information highway.

Internet Multicast

A technology called Internet Multicast may become the next World Wide Web (WWW) phenomenon.

Sometimes called the MBone (for Multicast backBONE), multicasting was invented by a then graduate

student at Stanford University, Steve Deering.
Whereas the World Wide Web (WWW) connects only two computers at a time, and then only for the few moments that information is being downloaded, the MBone continuously connects many computers at the same time. To use the familiar highway metaphor, for any one person the WWW only lets one car on the road at a time, and it must travel straight to its destination with no stops or side trips. By contrast, the MBone opens up streams of traffic between multiple people and so enables the flow of activities that constitute a neighborhood. Where the WWW ventures timidly to one location at a time before scurrying back home again, the MBone sustains ongoing relationships between machines, places, and people.
Multicast is fundamentally about increasing peripheral reach, derived from its ability to cheaply support multiple multimedia (video, audio, etc.) connections all day long. Continuous video from another place is no longer television, and no longer video-conferencing, but more like a window of awareness. A continuous video stream brings new details into the periphery: the room is cleaned up, something important may be about to happen; everyone got in late today on the east coast, must be a big snowstorm or traffic tie-up.
Multicast shares with videoconferencing and television an increased opportunity to attune to additional details. Compared to a telephone or fax, the broader channel of full multimedia better projects the person through the wire. The presence is enhanced by the responsiveness that full two-way (or multiway) interaction brings.
Like the inner windows, Multicast enables control of the periphery to remain with the individual, not the environment. A properly designed real-time Multicast tool will offer, but not demand. The MBone

provides the necessary partial separation for moving between center and periphery that a high bandwidth
world alone does not. Less is more, when less bandwidth provides more calmness.
Multicast at the moment is not an easy technology to use, and only a few applications have been developed by some very smart people. This could also be said of the digital computer in 1945, and of
the Internet in 1975. Multicast in our periphery will utterly change our world in twenty years.
Dangling String
Let's return to the dangling string. At first it creates a new center of attention just by being unique. But
this center soon becomes peripheral as the gentle waving of the string moves easily to the background.
That the string can be both seen and heard helps by increasing the clues for peripheral attunement.

The dangling string increases our peripheral reach to the formerly inaccessible network traffic. While
screen displays of traffic are common, their symbols require interpretation and attention, and do not
peripheralize well. The string, in part because it is actually in the physical world, has a better impedance
match with our brain's peripheral nerve centers.
In Conclusion
It seems contradictory to say, in the face of frequent complaints about information overload, that more
information could be encalming. It seems almost nonsensical to say that the way to become attuned to
more information is to attend to it less. It is these apparently bizarre features that may account for why
so few designs properly take into account center and periphery to achieve an increased sense of locatedness. But such designs are crucial. Once we are located in a world, the door is opened to social
interactions among shared things in that world. As we learn to design calm technology, we will enrich
not only our space of artifacts, but also our opportunities for being with other people. Thus may design
of calm technology come to play a central role in a more humanly empowered twenty-first century.
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speech interface versions and a qualitative analysis of their interactions towards those interfaces. Finally, speech interfaces implemented as information appliances (e.g., Amazon Echo, Google Home) may need to account for the preferences and needs of both children and adults [3]. As such, our third research question is (RQ3) How do the experiences and preferences of adults compare to those of children? We address this question by comparing both qualitative and quantitative aspects of children's interaction with our speech interfaces from those of 27 adults answering the same informational queries.

In the remainder of this paper, we begin by situating our work in the body of previous investigations on spoken dialog systems, personification and personalization in speech interfaces, and children's informational queries. We describe our methods, including the demographics of our participants, the speech interfaces used, and the procedure of the investigation. We discuss our analysis and findings in response to each of the above research questions. Finally, we conclude with a discussion that provides specific implications for the design of future speech interfaces for homes.

RELATED WORK

We outline previous work in spoken dialog systems and speech interfaces for children and adults to situate the three research questions we are exploring in this investigation.

Spoken Dialog Systems & Speech Recognition

Spoken dialog systems and speech recognition systems are becoming more and more sophisticated, with systems like Apple Siri, Amazon Alexa, Microsoft Cortana, and Google

Assistant available as off-the-shelf products. Recent advances in the field focus on personalizing to the needs of specific users [6], maintaining an understanding of context over multiple turns of interaction [8], and asking appropriate clarifying questions to guide the user [30]. However, most of these advances target adult users.

A number of papers provide evidence that speech is a promising mode of computer interaction for children (e.g., [41]). Proposed applications of such systems are as diverse as play/entertainment (e.g., [5,22,28,32,33]), social skill practice (e.g., [15]), literacy education (e.g., [21,39,45]), and in-car entertainment (e.g., [18]). However, spoken dialog systems for children is an area with significant potential for improvement. First, speech recognition with children is notoriously difficult (e.g., [17,26,50,55]), especially for spontaneous utterances which have been described as "disfluent and ungrammatical" [21]. Second, recognizing intent is significantly more difficult than just recognizing speech, even with adults [53]. While several investigations have sought to generate datasets of children's speech (e.g., [29]), datasets of children's spontaneous utterances are lacking [41]. Our work begins addressing this gap, however we are not seeking to improve the state of the art in speech recognition or spoken dialog systems, but rather answer several open questions in speech interface design.

Personification & Personalization in Speech Interfaces

One open question in speech interfaces for the home is the role of personalization and personification. HCI researchers have pursued personalized and personified interaction with speech interfaces since the early 1990s [38], though whether these kinds of agents lead to the best user experience remains an open question even after decades of investigation (e.g., [52]). For example, one previous study found that personified agent-like output from a speech device led adults to interact with it in ways that were not well supported by their system (e.g., asking the device direct questions) [24]. Adults interacting with Siri reported negative responses to a similar mismatch between the level of personification and actual capabilities of the speech interface [37]. An investigation of adults' interaction with a virtual reception-

ist found that people attributed different amounts of personification to the receptionist and interacted differently with it based on that attribution [31]. Another investigation focused on voice output found no effects of output "embodiment" (speech coming directly from smart objects vs. a disembodied home control agent vs. a home controlled agent that was also embodied as an on-screen avatar) on users' experience, though the disembodied voice was slightly preferred overall [47]. Nonetheless, most current commercial speech interfaces (e.g., Siri, Amazon Alexa, Google Assistant, Cortana) are both personified and personalized. The role of the user's age in personification and personalization preferences is even less clear. One paper suggested that "factual" versus "social" interaction with a dialog system may be an inherent user preference rather than an age-dependent characteristic [56]. Other preliminary work with speech interfaces in the home shows that children interact with voice systems differently than adults. For example, children are more likely to include social exchanges with the system (e.g., "bye!") [4]. Older participants are more likely than younger children to have a negative affective response to speech interfaces that violate privacy expectations (e.g., know information that the child didn't explicitly tell them) [34]. Outside of speech interfaces, at least a few investigations have suggested that personified search agents may help children interpret query results and found that this approach worked best with 8- and 9-year-olds (older children found it to be too "childish") [20]. Systems can be fairly accurate at distinguishing between adult and children's speech [43], however little is known about how a system could then adjust its personification and personalization in the most appropriate way to ages or preferences of users. Our work addresses this gap, with the goal of leading to more tailored speech appliances for families (e.g., as suggested in [3]).

Children's Informational Query Practices

While there has been substantial previous work on speech interfaces for children's entertainment (e.g., [18,22,32,33]), we focus on informational queries as the specific context of interaction with speech interfaces. There are two reasons

for this decision. First, these interactions represent an im-

portant use case for speech interfaces. Previous work revealed that the plurality (30%) of family speech interactions with a home voice kiosk [4] and the plurality (45%) of children's interactions with Apple's Siri [36] fell into the information seeking and web search categories. Second, there is substantial potential for leveraging speech interfaces to address a number of challenges that children face with current text-based search practices. Log analysis has found that children's web searches are frequently "unsuccessful" and "confused" [13,14]. The major challenges identified in previous investigations of children's web search include spelling, typing, and query formulation [10,11]. Spelling and typing difficulties may be amplified by children's tendency to use longer, natural language queries [12,27] (though this may be culturally dependent, as a study of German children's search found that they were more likely to have shorter queries [19]). Speech interfaces may remove the barriers of spelling and typing, allowing children to focus on the task of query formulation.

Query reformulation in response to a misrecognition or misunderstanding is an important aspect of interacting with speech interfaces. This has been found to be a challenging task even for adults, who employed strategies as diverse as word substitution, phrase re-ordering, and phonetic emphasis [25,37]. Other previous work on speech-based home automation controls investigated adults' responses to different types of errors (i.e., ones leading to stagnation, regression, or partial progress towards a goal) [47], again reiterating that reformulation may be challenging. It may be even more difficult for children, but little is known about children's practices with speech interface query reformulation. Previous exploratory work that has examined children's interaction with Apple's Siri through a content review of YouTube videos found that children had substantial trouble dealing with speech interface errors, relying mostly on phonetic emphasis in reformulation [36]. Additionally, in a study with a similar WoZ design to ours, Oviatt et al. found that children change the prosodic (e.g., speed, pitch) quali-

ties of their speech to match an animated agent's [48], but did not investigate semantic reformulations or adaptations. One of our goals in this work is to understand how children deal with restating or restructuring queries when they cannot get to an answer.

METHODS

In this section, we describe our setting, participants, and detail the system setup and procedure to support replication.

Setting

The study took place in the research outreach building at the Minnesota (MN) State Fair. The Driven2Discover building is a permanent facility on the State Fairgrounds, visited daily by thousands of fair attendees who are representative of the population in the Minnesota.1 The permanent facility provided a relatively private and quiet space

1 http://d2d.umn.edu/

for the study to take place (e.g., other studies were separated by curtains and the building was protected from the general hustle and bustle of the Fair). There were two study stations set up in the building booth (see Figure 1), as well as another station for gathering initial information and consent from the children and parents.

Participants

We recruited child participants (ages 5–12), along with a parent or guardian who could provide consent, give information about the child, and potentially participate in this study. One benefit of our chosen study setting was the ability to recruit and include a greater diversity of families than most lab-based investigations. Families passing through the research facility and choosing to participate in the study were representative of Fair attendees as a whole. For example, 25% of the visitors were from rural counties (consistent with state population) and 28% of the parents did not have a college degree (consistent with published demographics statistics from the 2016 State Fair).

During the course of the study, 87 children completed the procedure (57% female; M = 9 years old, SD = 1.99). Parents or guardians were given one of three options:

1. If there was an empty station, they could attempt the tasks themselves (otherwise, we prioritized children).

2. They could sit next to their child to help read the questions and write answers (any assistance given by parents beyond reading and writing help, such as hints or prompts, was logged and included in our analysis).
3. They could do neither and wait for their child to complete the study.

Given these options, 27 adults completed option one, providing us with a base of comparison for adult participants (48% female; M = 47 years old, SD = 10.61). All

Figure 1. A child points to one of the interfaces in the study setup. Each interface is represented as a plastic bin with a speaker. The Wizard-of-Oz (visible behind the cardboard divider) controls the text-to-speech output of each device.

participants (adults and children) were either native English speakers or agreed with the statement "I am comfortable speaking English." We also requested parents to tell us which (if any) voice assistants they and their children have used in the past (e.g., Apple Siri, Amazon Alexa). 96% of the adults and 94% of the children had used one or more different voice assistants in the past. Parents were more likely to have used these interfaces infrequently (mode response was "less than once a week"), while children were more likely to use these interfaces frequently (mode response was "multiple times per day").

Systems

To address our research questions, we developed multiple versions of a voice assistant. We discuss the physical hardware employed, the software and Wizard-of-Oz training, and the specific modifications made in each condition.

Equipment

Each study station housed three variations of speech interfaces, each represented as a different plastic housing and an AISBR 3W wired speaker (see Figure 1) to allow participants to more easily distinguish between and refer to each interface. All participants were audio recorded using a Blue Yeti USB microphone, which contains a tri-capsule array to support field recording of human voices at a 48kHz sample rate and 16bit bit rate. All of these peripherals were connected to a laptop running the Wizard-of-Oz's control soft-

ware and Audacity recording software.

Wizard-of-Oz Controls and Training

To focus our investigation on the role of question reformulations (rather than recognition factors), we chose to use a Wizard-of-Oz (WoZ) technique to simulate a voice assistant. This allowed us to have human-quality speech recognition, removing this confound from the investigation. As in other WoZ studies (e.g., [9]), to allow the Wizards to provide near real-time response to the participants, we had to develop custom WoZ control software. The team created a Python GUI (see Figure 2) to allow the Wizard to quickly and consistently select common responses and statements, as well as directly edit response text when modifications were necessary. The Wizards followed a specific script to ensure that participants received consistent responses from the system. Response text was converted to speech using Microsoft's Zira voice (female voice with an American accent) and the pyttsx Python text-to-speech library.

Five Wizards supported this investigation, allowing work to occur in shorter shifts to avoid inconsistencies due to fatigue. All Wizards used a common protocol and guide for contingency responses and trained together through multiple piloting sessions to increase consistency in Wizard response. Wizards were visible to the participants, though a privacy screen hid their immediate actions (see Figure 1). Participants were told that the Wizards were there to help the voice assistants. Due to our significant GUI shortcuts and piloting, most responses were quick and required no typing, revealing little about the Wizard's role. Due to the increasingly common use of human computation as a technique in computing systems, the IRB did not view WoZ to be an example of deception so no debriefing was required with participants who did not inquire about the specific functioning of the system. Only two adults expressed suspicion that the researcher had a greater role and they were debriefed after the study. None of the children expressed suspicion or inquired about how the system worked.

Three Conditions

To answer our questions regarding the role of personification and naming personalization, we built three variations

of the speech system (supplementary materials include examples of full scripts of interaction with each system):

• Voice Search System – the non-personified and non-personalized system never referred to itself in first person, gave only task-related responses, and did not mention the participant's name, e.g.: "Welcome to the voice search system. Please, say your question."

• Fraga2 – the personified and non-personalized system referred to itself in first person, gave some responses that were not task-related, but did not mention the participant's name, e.g.: "Hello, I am Fraga. Do you have a question for me?"

• Swali2 – the personified and personalized system referred to itself in first person, gave some responses that were not task-related, and periodically referred to the participant's name and age, e.g.: "Hello Jake, I am Swali. I see you are 6 years old. That makes you 5.5 years older than me. Do you have a question for me?"

2 "Fraga" and "Swali" mean "question" in Swedish and Swahili, respectively.

Figure 2. A custom software interface guided the Wizard. Many common interactions (e.g., greetings, hints) were pre-programmed. Ad-hoc interactions could be typed directly into the response box or edited from an existing scripted response. After a participant stated their question, a "ding!" sound provided feedback that the "system" heard it. The interface guided the Wizard through condition order and logged interactions.

Given the limited nature of children's attention spans and that reliable preference measures for children require within-subjects comparisons [57], we minimized the number of conditions. We omitted the non-personified, personalized condition as least natural and least interesting.

The Wizards were directed by the GUI as to the order of these interfaces and would switch speakers appropriately. The control interface automatically applied interface-specific wording and formatting to the dialog (e.g., adding the name to the greeting in the personalized condition). We chose to use the same voice for all three variations of the

interface in order to avoid user preference of certain voices or dialects from influencing their selection.

Procedure

Potential participants were solicited by a research team member as they passed through the Driven2Discover Building. If they were still interested in the study after the brief pitch, the families were led to a table where the study was explained in more detail and a researcher answered their questions. The child wrote their first name on an assent form, if able. The parent filled out consent for himself or herself and for the child to participate, as well as a short demographic and background questionnaire.

The study represents a within-subjects design, with each participant asking at least one question to each interface, in counterbalanced order. It is important to note that we had specific ethical considerations and constraints that took priority over procedural consistency in certain cases. In our discussions with our IRB, it became clear that it was important to make sure that every child left the study feeling that they "succeeded" at the assigned task. This consideration led to four study design decisions. First, members of the family could sit next to the participant, potentially offering advice (this was used to allow siblings under 5 to serve as "special helpers" and get a toy at the end). Second, if a child was not making any progress towards a question (e.g., continuing to say it the same way), they were offered progressively more significant hints by the system or by a researcher. Third, if more than three minutes passed without a child arriving at an answer to a question or the child was getting increasingly dejected, the Wizard would "lower the bar," giving an answer in situations where the protocol would otherwise require the system to request additional clarification. Fourth, there were two "levels" of questions (with the second question labeled "bonus"). If a participant was able to arrive at an answer to the first question within one minute of using the system, they were presented with a harder bonus question for the same interface.

The questions were presented to participants on a sheet of paper and they were asked to write a response once they arrived at an answer with the voice systems. The easier first

question typically provided some context and then a question that referred to that context (see Table 1). To arrive at an answer, participants had to ask their question in a way that integrated the provided context. For example, for the second question in Table 1, the participants had to ask the system about the number of mini-donuts sold in a particular year and compare the amount to the one stated in the question. To answer the more difficult "bonus" questions, the participants had to decompose a given question into two parts. For example, for the first "bonus" question in Table 1, the participant had to first ask what the newest ride was and then ask where that ride was located at the fair. The six questions were always presented in the same order, however the order of the interfaces used was counterbalanced (Latin square). This was done to control for the role of both

Table 2. Descriptive statistics of the comparisons between conditions for children and adults. Average hints and exchanges were calculated across the first question in each condition (as that was the one that was completed by all participants).

Single Condition Results	Grouped by Personification		Grouped by Personalization	
Voice Search Fraga Swali	Non-Personified	Personified	Non-Personalized	Personalized
Children Preferred by # (# expected if random)	17 (27)	34 (27)	31 (27)	17 (27)
	65 (55)	51 (55)	31 (27)	
Avg. Hints	0.87 0.96 1.02	0.87 0.99	0.95 1.02	
Avg. Exchanges	2.73 2.85 3.10	2.73 2.98	2.79 3.10	
Got to Bonus?	33% 32% 26%	33% 29%	33% 26%	

Adults
Preferred by
(expected if random)
5
(7)
6
(7)
11
(7)
5
(7)
17
(15)
11
(15)
11
(7)
Avg. Rating 3.63 3.40 3.80 3.63 3.60 3.52 3.80
Avg. Hints 0.04 0.12 0.12 0.04 0.12 0.08 0.12
Avg. Exchanges 2.00 2.46 2.23 2.00 2.35 2.23 2.23
Got to Bonus? 85% 61% 75% 85% 68% 73% 75%
Table 1. Questions and “bonus” questions given as tasks.
Three Initial Questions
The biggest Pig in the history of the MN State Fair was Reggie the Pig in 2010. How much did he weigh?
500 thousand corn dogs were sold at the MN State Fair in 2016. Were more or fewer mini donuts sold there that year?
The oldest ride at the MN State Fair is “Ye Old Mill.” What year was it new to the MN State Fair?
Three “Bonus” Questions
Where can you find the newest ride at the MN State Fair?
Did more people attend the MN State Fair in 2015 or 2016?
Which State Fair is older, the MN State Fair or TX State Fair?
order effects and natural variations in question difficulty on participant preferences.
After each question, children were asked to rate the voice system they used on the smiley-o-meter scale [51]. As suggested in previous work, we used this scale as an opportunity for children to pause and reflect on the experience rather

than as a reliable metric of preference [51,57]. Adults specified their ratings on a similar 5-point scale. After trying the three interfaces, we asked all participants to pick their favorite interface of the three (a three-way variation of the "This or That" method [57]), explain why they liked it, and ask it one other question on any topic. All questions and sections of the study were optional—for each question, we only report results for the subset of participants who answered it. At the end of the study, parents and children were given a choice of a university-branded drawstring pack or a small stuffed animal as compensation for their time.

RESULTS

In this section, we describe the analysis and discuss our findings to each of our three guiding research questions.

RQ1: How do children restructure informational queries towards a speech interface?

We reviewed the logs and audio recordings of all the participants, coding the exchanges initiated by each participant to answer each question. Since our IRB required that all children arrive at an answer by the end of each interaction, we had to use an alternative measure of effectiveness than answer accuracy. To estimate this effectiveness, we coded the number of hints and/or prompts participants required to reach an answer (which could come from the system, the researcher, or the parent when the child was stuck). We also noted when a child attempted the bonus question (meaning that they arrived at the answer to the first question within one minute) as a signal of success with the interface. Table 2 provides the descriptive statistics of each of these measures across conditions. Generally, children struggled with the reformulation task, requiring one or more hints and with less than half of the children reaching the "bonus" question. These difficulties reduced with age, with the chance of getting to the bonus question significantly increasing ($r = 0.30$, $p = 0.006$) and the number of needed hints significantly decreasing ($r = -0.33$, $p = 0.004$).

A team of four researchers reviewed the recorded audio of each interaction, taking notes, transcribing (example transcripts available in supplementary materials), and describing each instance of a reformulation. Thus four researchers

transcribed, memoed, and open coded each transcript individually following the process described by Lofland et al [35]. Then the four researchers took part in a workshop led by the lead author to arrive at clusters of codes through multiple rounds of constant comparison, using "abductive" analysis [40] to develop our categories. Once our codebook was developed, we each reviewed all the transcripts again, applying those codes to the original dataset. The following categories of reformulations emerged through this data-driven inductive process:

Off-Course – changing the question to something relevant to the topic that the system can answer, but that does not get the asker closer to the target answer. Examples:

Child repeatedly asks "Was Reggie the Pig the fattest pig in 2010 at the MN State Fair?" receiving "Yes" as a response from the system but not knowing how to follow up about the pig's weight.

Child gets frustrated with repeatedly getting the response "That question is too complicated for me" and changes the topic, asking the system: "Who is the strongest superhero?"

Restating or Repeating – changing how a question is pronounced or emphasized, without changing any words or structure of the question. Examples:

System asks the child to "Please, ask the question in a different way." Child sings the question to the system.

System asks the child to "Please, ask the question in a different way." Child repeats question louder.

Substituting Words – changing a word or phrase in a question without adding any additional information or removing any complexity from the previously stated question. Examples:

Child rereads the whole question about which State Fair is older, substituting "which is younger" for "which is older."

Child rephrases question as "How many pounds was Reggie the Pig?"

Reordering – reordering the components of a question without adding any additional information or removing any complexity. Examples:

Child rephrases question as “True or False: the MN State Fair is older than the Texas State Fair.”

Table 3. Prevalence of categories of query reformulation by age. We excluded 3 children whose audio data was incomplete (one or more condition was inaudible) and 5 children who did not make any independent reformulations (all reformulations were prompted). In this study, starred categories helped participants get closer to an answer; others were not effective.

	Children, 5–7 (N = 19)	Children, 8–12 (N = 58)	Adults (N = 27)
Off-Course	5%	9%	0%
Restate	53%	64%	30%
Substitute Words	53%	52%	63%
Reorder	37%	53%	37%
State Context	26%	22%	30%
Expand Pronouns*	32%	62%	70%
Add Context*	58%	66%	44%

Child rephrases question as “How many fewer mini-donuts were there sold than corn dogs in 2016?

Stating Context – adding additional information before asking a question, but phrasing this addition as keywords or statements (not integrated into the question). Examples:

Child states “Reggie was the fattest pig in 2010 at the MN State Fair. How many pounds was he?”

Child states “The Great Big Wheel is the newest ride at the MN State Fair. Where is it located?”

Expanding Pronouns in Question – replacing the non-specific pronoun in a question with a specific noun. Examples:

Child expands the pronoun “he” to “How much did Reggie the Pig weigh?”

Child expands the pronoun “it” to “What year was Ye Old Mill first introduced?”

Adding Context Phrases – adding context following or preceding the noun in a question to narrow to the specific

case of interest. Examples:
Child asks the questions one at a time until the system is able to answer: “How much did Reggie the pig weigh?” “How much did Reggie the pig from the State Fair weigh?” “How much did Reggie the pig ... from 2010 ... from the fair ... from the MN State Fair weigh?”
Child first asks about the number of mini donuts and corn dogs without specifying a year or location. System asks him to be more specific. Child expands: “At the MN fair were there ... in 2016 ... were there more or less mini donuts sold than corn dogs?”
It is important to note that in this study, the first five types of reformulations did not help the participants arrive at an answer. However, when interacting with other speech interfaces, some of these may be helpful strategies. Table 3 reports the percent of children who employed each reformulation. The most common unsuccessful reformulations were restating without changing the question (53% and 64%, among 5–7 year-olds and 8–12 year-olds respectively) and changing the words in the question without changing its structure (53% and 52% in the two age buckets). The most common reformulation strategies that got the participant closer to the answer included adding context into the question (58% and 66% in the two age buckets) and expanding the pronouns in a question (32% and 62% in the two age buckets). Most children tried a number of different strategies (on average, three or more), but many were stuck in a single strategy until a hint or prompt was provided.
After the children had the opportunity to use each interface, they could ask one more question on any topic to the interface of their choice. We categorized these based on the structure and assumed intent of the question. While children were told that they could ask a question about anything at all, many seemed to be quite influenced by the previous tasks (this was expected given previous preliminary work [53]). 32% of their questions were quantitative questions about other state fair topics (e.g., “What year did the Ferris wheel open?”) and another 34% were quantitative questions unrelated to the state fair (e.g., “Are there more dogs in the world than cats?”). Qualitative questions (e.g., “How do

staplers work?") accounted for 8% of the dataset. Two other interesting categories emerged. In 18% of the questions, children wanted to learn more about the experiences and interests of the interface (e.g., "What is your favorite football team?"). In 9% of the questions, children tried to test the interface (e.g., "What color bucket is under you?" or "What is zero divided by zero?").

RQ2: How does a speech interface's personification and naming personalization affect children's experience?

We asked each child to interact with all three of the interfaces, thereby gauging their response to personified and personalized systems. Table 2 provides descriptive statistics of metrics gathered in each of the conditions. We compared children's preferences for personified vs. non-personified conditions using Chi-Square Goodness-of-Fit test, finding that children did indeed prefer personified interfaces at a greater and statistically significant rate ($p = 0.015$). A similar comparison between personalized and non-personalized conditions was not statistically significant. We did not observe a relationship between the child's age and their interface preference, though it is possible that one could emerge in a study with more participants of each age.

Effectiveness measures were similar across the conditions. When compared using a Repeated Measured ANOVA, the difference in the number of hints required and exchanges made across conditions was not statistically significant. Similarly, comparing the number of children getting to the bonus question in each condition versus an even distribution across conditions using the Chi-Square Goodness-of-Fit test did not show a statistically significant difference. Therefore, we conclude that personification and naming personalization do not influence children's effectiveness with speech interfaces in a statistically significant way.

We asked each child about why they liked a particular interface best. Of those that could give an answer beyond the tautological (e.g., "I liked it better"), 62% said that they thought that their favorite interface understood them better or was less confused about their questions (this was mostly an order effect—children developed more effective reformulation strategies by the third condition). Eleven percent

of the answers were related to irrelevant surface consideration (e.g., favorite color), however removing these responses from the analysis did not change the direction or magnitude of the result. Two children mentioned liking the personalized interface best because of its personality ("more friendly" and "polite") and another four children explicitly mentioned liking the interface that "knew my name." On the other hand, another four children were explicitly turned offCommunication, Emotion & EngagementIDC 2018, June 19–22, 2018, Trondheim, Norway306 by the naming personalization saying that it was "creepy." One of these children loudly exclaimed, "Are you stalking me?!" when the system referred to him by name and age. However, it is important to interpret this reaction in context—this information was solicited by the researcher not directly by the robot, so this reaction may be moderated in other arrangements. Fewer children provided interface-specific reasons for liking the non-personified one, though one did mention liking it because it "spoke faster and was more efficient." While the qualitative experience for this individual was more efficient with the non-personified interface and there was a descriptive difference between the conditions in this direction, it was not statistically significant so it does not generalize across our sample.

When given the opportunity to ask their favorite interface any question they wanted, it was interesting to note that questions about the interests and experiences of the interface and qualitative questions were directed almost exclusively (all but one) towards the personified conditions.

RQ3: How do the experiences and preferences of adults compare to those of children?

As can be expected, adults faced fewer struggles with question reformulations, with only a few requiring hints and with the majority reaching the bonus question (see Table 2). We did not see any examples where adults went so off-course in their reformulations that they asked questions that did not get them closer to the answer (though in two cases, adults asked additional irrelevant questions after completing their task). Fewer adults (30%) than children (53% or 64%, depending on age) focused on the unsuccessful strategy of simply restating the question with a different emphasis or

pronunciation. Adults employed some successful strategies more often than children. For example, older participants were more likely to expand pronouns (32% of 5–7-year-olds, 62% of 8–12-year-olds, and 70% of adults). They were less likely to reformulate the question with additional context, but only because many included all the necessary context phrases from the first formulation.

Adult preferences for naming personalization and personification were not statistically significantly different from random when tested with the Chi-Square Goodness-of-Fit test ($p = 0.095$). There was no statistically significant difference between their ratings of each condition on a 5-point Likert-type scale when compared with a Repeated Measures ANOVA. Like the children's results, there was no statistically significant differences between conditions for adults' effectiveness with the systems in terms of the number of hints required, number of exchanges, or the likelihood of getting to the bonus question. While descriptively, adults seemed to prefer the naming personalization condition at greater rates than children, the difference in preference distributions between adults and children was not statistically significant when compared with a Chi-Square test.

Fourteen of the 22 adults who had a preferred interface provided a reason for that selection. Interestingly, the most common (36%) reason explicitly referred to liking the "personality" of the naming personalization interface best, describing it as "like a friend," "witty," and "more personal." However, one adult did mention being "creeped out" and disliking the same interface for saying their name (even though they understood that they provided their name to the system). One adult also mentioned that they liked the non-personified interface best because of its efficiency ("not a lot of extra talking"). Finally, adults generally recognized that all three interfaces understood them equally poorly and were more likely to reflect on the role of order in their system preferences, acknowledging that they became better at asking questions by the end of the study.

While almost all children took the opportunity to ask their favorite interface another question, only 13 out of the 27 adults did so. As with the child participants, 38% of these

were quantitative questions related to the state fair and another 38% were quantitative questions on other topics. None of the adults asked qualitative questions or attempted to learn more about the experiences and interests of the interface. However, as children did, 23% of the adults did try to ask questions that tested the interface, usually by asking a question where they already knew the answer (e.g., "Who was the first president of the United States?").

DISCUSSION

In this section, we discuss the implications of our findings on design and research in speech interfaces for families.

Limitations and Future Directions

All study designs have inherent limitations. We had to make specific decisions regarding the wording of questions, responses, and specific operationalization of concepts like "personification" and "naming personalization." Children may have personified the non-personified condition despite the language used by the system, as previous work shows that personal pronouns are not necessary for perspective-taking [2]. Similarly, personalization can be much more nuanced and useful than merely referring to a person by their first name [16]. For example, it would be a useful personalization to use the participants' location to provide context for the questions or to tailor the systems responses based on a child's ability. Given that many of the concerns children expressed against personalization were privacy-based, the privacy theory of "proportionality" [23] suggests that some of those concerns would be ameliorated if the personalization provided a functional benefit in interpreting and answering questions. Also, it may not have been transparent to the child how the systems learned their name, since these were entered by the researchers. But, this parallels the real-world situation where a parent would generally provide information for a child's voice assistant account. Certainly, we do not consider our investigation to be the final word on personalization of speech interfaces, but rather a point of evidence towards the idea that a parent entering a child's name into a speech system does not support a personalization that provides measurable value to children.

We made study design choices that allowed us to collect data from a large, diverse sample in a field setting. While this has many advantages, it also introduced several limitations. First, our study was not tightly controlled or in a lab setting. This may mean that there were times when siblings, parents, etc. distracted the children from their tasks. However, this also represents a more ecologically valid situation. Second, we provided most of the questions asked by the participants. We selected questions that were challenging and that would necessitate reformulation (similar to the approach of previous work on children's online search, e.g., [9]). We picked questions that were relevant to the setting and similar in style to the kinds of questions children may be asked to answer on a homework worksheet or similar assignment. However, we do not know to what extent these compound complex questions appear in the lexicon of children's interactions with speech interfaces in the wild. Finally, each participant had a relatively brief interaction with a speech interface that belonged to the researcher. There are two possible biases that this introduced in the dataset. First, we could not observe long-term learning effects. We did observe that participants learned from earlier interactions and their performance improved even in the short time of the study. It is important to conduct a follow-up study with a home or mobile speech device in the wild, to understand which reformulations remain problematic in the long run and which are quicker adaptations. Second, privacy concerns may have been exacerbated by interacting with a speech assistant that belonged to the researcher, rather than a personal device. It is possible that some of the "creepiness" factor of naming personalization may have been ameliorated if participants interacted with their own personal speech device. This was not possible in the context of our controlled investigation, but would be an important and promising follow-up study if participants could be given personal devices to keep and interact with long-term.

Considerations for User-Friendly Speech Interfaces

Observing the struggles and relatively low success rate of our child participants, we reflect that current speech interfaces may be considered poor designs from a classic design

perspective (e.g., [46]). They provide no constraints on the kinds of statements that may be directed to them, leading to the observed high number of exchanges before arriving at an answer. They provide little feedback regarding what they heard and understood, leading most children to spend significant time on phonetic reformulations. They give almost no visibility as to why the system may be struggling with a particular question (e.g., Amazon Alexa simply answers "I don't know" if it is confused by any part of the question). Many of our participants struggled but took reasonable paths—first assuming that a question was misheard, then assuming that a word was not understood, before moving on to more effective strategies. However, the process was undeniably frustrating and many of the children required help to get past the first two strategies and to arrive at questions that would yield an appropriate answer.

Currently available systems rarely (if ever) provide hints or clarification. Based on each of the common reformulation types and the types of hints and prompts that were most useful to the participants in our study, we recommend that systems integrate the following types of clarifications:

• Restate what was heard if the system fails to meet a confidence threshold. Children are used to being misheard by speech interfaces and most of them focused on phonetic reformulations. With feedback that the system heard the question correctly, they may proceed to semantic reformulations.

• If a particular piece of context is needed but missing (e.g., year, identity of a particular pronoun), follow up requesting this information (e.g., "What is "it" in your question?). It may also be reasonable to make a "best guess" about missing information from previous questions or by assuming current location, year, etc.

• If a part of the question is known but the entire question is too complex, provide some information on what is known and request clarification for the rest (e.g., "I know that the newest ride at the fair is the 'The Great Big Wheel,' what would you like to know about it?").

• If a particular style of question is difficult for the system, clarify this and provide guidance (e.g., "Compari-

sons are hard for me. Can you ask about each part of your question separately?").

Additionally, sometimes a significant amount of contextual information is needed to provide the correct answer to a question. It can be difficult and awkward for children to construct complex questions that integrate all of the necessary components. While many children (58%-66%) eventually attempted integrating contextual information directly into the question, they struggled with the compound sentences that resulted, pausing after each informational element. We also saw that stating information up-front rather than integrating it directly into the question was a strategy employed by both adults (30%) and children (22%-26%). Speech systems could become more usable in that regard by allowing contextual information for a particular question to be provided in the form of a sentence or multiple sentences. For example, instead of having to ask a system "What will the weather be like this Thursday and Friday in Trondheim?" it may be easier for users to give upfront context ("I am traveling to Trondheim on Thursday.") followed by the particular question ("What will the weather be like?").

Costs and Benefit of Personifying & Personalizing

Previous work in the field of speech interfaces had provided divergent findings regarding the potential roles of personification and personalization. This investigation helped address some aspects of this open question.

None of the participants had specific objections to personification of interfaces, though several participants did note that non-personified interfaces covered less non-task content and thus were more efficient at quickly providing an answer. Children responded to personification, preferring personified conditions to the non-personified. Children also seemed more willing to direct a broader variety of questions, including qualitative ones, to personified interfaces. Whether because of actual confusion or as a playful act, children asked the personified interfaces question about their experience and preferences (echoing previous studies [3]). This may not be entirely positive, as it is difficult to balance engaging in the play to provide a "fun" answer and

answering questions honestly without misleading the child as to the nature of the interaction (e.g., if asked about its favorite food, should the system lie or should it say that it does not eat?). It is worth noting that the same consideration need not be extended to adults—we saw no examples of adults asking these kinds of questions. While not stated as a concern by any participants, researchers in other domains have noted that increased personification of interfaces may create a "robotic moment," in which children may become confused the role and agency of humans compared to machines [54]. This may be a trade-off of personification if empirically confirmed in future research.

We did not find statistically significant evidence favoring naming personalization as operationalized in our study. One common objection cited by several children and one adult was that it was a violation of privacy expectations to have the speech device know specifics like their age and name, even though they themselves provided this information to the researcher. This objection complements previous investigations of children interacting with robotic agents [34]. On the other hand, some children and adults did find the personalized system to be more friendly, witty, and polite. This may be an individual difference. However, it is also worth noting that even the kind of surface naming personalization mentioned in this study may not be easy to accomplish in the field. First, it may require interfaces to distinguish between users based on voice, which is possible but not trivial [43]. Second, it is increasingly common for children to have names that are non-traditional, unusually spelled, or names from non-western cultures. This increases the likelihood that a speech system will mispronounce a child's name, which may reverse perceptions of friendliness and politeness. Given that benefits are not evidenced in this investigation, naming personalization should not be a high priority feature. However, it is again important to reiterate that there are many other possible types of personalization beyond "naming," which may provide a different set of costs and benefits to the participants.

Building for the Whole Family

One of the surprising descriptive findings of our study is

that 93% of children had used one or more speech interfaces prior to the study and the plurality of these children used such interfaces multiple times every day. While we thought of this interface modality as "emerging," it became clear that it was already well-entrenched in the lives of children. There are three ways children typically interact with speech interfaces. Some children may have their own smart phone or tablet. In this case, it may make sense to default to a personified interface, letting the child control their level of personalization. Other children may experience so-called "pass-back"—the opportunity to interact with speech interfaces on their parents' smart phone. It may be useful for such devices to detect children's voices (e.g., [44]) and remove any personalization features aimed at the parent while retaining personification. Third, children may live in a home that has a household speech information appliance (e.g., Google Home). Such devices need to account for the specific preferences of multiple users [2]. Our study complements previous investigations showing that interface personification preference may be more related to individual differences than age [56]. We did not see specific age effects found in other previous work (e.g., older children considering personification to be too childish) [20]. There was a substantial split regarding personalization and there were also a fair number of participants who preferred the non-personified condition (i.e., a family of four is not unlikely to have one person in this category). It is an open question whether it would provide the best experience for the whole family to adapt to individual preferences or whether families would prefer to have a consistent experience even if some individuals' preferences are violated.

CONCLUSION

Most children in our study used voice assistants and many of them used such interfaces multiple times every day. Informational queries are a common use case, but prior to this investigation little was known about how children formulate questions towards speech interfaces. Our work addresses this gap by observing 87 children asking questions of three variations of Wizard-of-Oz speech interfaces. We found that children struggled with reformulating questions,

with most of them requiring hints to complete the task. Though most children eventually tried effective reformulations such as substituting objects for pronouns and providing context within the question, many children first began with surface reformulations such as repeating the question or substituting synonym words. Older children and adults were more effective than younger children at informational query reformulation. By comparing variations of the interface, we discovered that children preferred personified interfaces, but showed no preference towards the interface that was personalized with their name and age. We suggest several considerations for future speech interfaces. First, personified interfaces are well indicated, while naming personalization (especially, when that name is provided by others) is not. Second, we point to five strategies to support more effective reformulations: providing feedback on what was heard, asking for missing context, clarifying what is known, specifying formulations that are difficult for the system, and allowing context to be provided as a statement.

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SELECTION AND PARTICIPATION OF CHILDREN

This study was reviewed and approved by a University IRB. We invited families with children between the ages of 5 and 12 to participate at the MN State Fair event. A researcher explained the study and its risks to both the child and parent and answered any questions. If interested, parents signed informed consent and a parental permission form. A researcher read a paper assent form out loud to each child. If assenting to the study, the child wrote their name on the form (as able). Both parental permission and the child's explicit assent were necessary for the study to proceed and either person could ask us to stop the study at any time (the child still received a toy for their help).

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Moving on from Weiser's Vision of Calm Computing: Engaging UbiComp Experiences

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Abstract. A motivation behind much UbiComp research has been to make our lives convenient, comfortable and informed, following in the footsteps of Weiser's calm computing vision. Three themes that have dominated are context awareness, ambient intelligence and monitoring/tracking. While these avenues of research have been fruitful their accomplishments do not match up to anything like Weiser's world. This paper discusses why this is so and argues that is time for a change of direction in the field. An alternative agenda is outlined that focuses on engaging rather than calming people. Humans are very resourceful at exploiting their environments and extending their capabilities using existing strategies and tools. I describe how pervasive technologies can be added to the mix, outlining three areas of practice where there is much potential for professionals and laypeople alike to combine, adapt and use them in creative and constructive ways.

Keywords: calm computing, Weiser, user experiences, engaged living, UbiComp history, pervasive technologies, proactive computing.

1 Introduction

Mark Weiser's vision of ubiquitous computing has had an enormous impact on the directions that the nascent field of UbiComp has taken. A central thesis was that while "computers for personal use have focused on the excitement of interaction...the most potentially interesting, challenging and profound change implied by the ubiquitous computing era is a focus on calm." [46]. Given the likelihood that computers will be everywhere, in our environments and even embedded in our bodies, he argued that they better "stay out of the way" and not overburden us in our everyday lives. In contrast, his picture of calm technology portrayed a world of serenity, comfort and awareness, where we are kept perpetually informed of what is happening around us, what is going to happen and what has just happened. Information would appear in the centre of our attention when needed and effortlessly disappear into the periphery of our attention when not.

Now regarded as the forefather of UbiComp, Weiser has inspired governments, researchers and developers across the globe. Most prominent was the European Community's Disappearing Computer initiative in the late 90s and early 2000s, that funded a large number of research projects to investigate how information technology could be diffused into everyday objects and settings and to see how this could lead to

new ways of supporting and enhancing people's lives that went above and beyond what was possible using desktop machines. Other ambitious and far-reaching projects included MIT's Oxygen, HP's CoolTown, IBM's BlueEyes, Philips Vision of the Future and attempts by various telecom companies and academia to create the ultimate 'smart home', e.g., Orange-at-Home and Aware Home. A central aspiration running through these early efforts was that the environment, the home, and our possessions would be aware, adapt and respond to our varying comfort needs, individual moods and information requirements. We would only have to walk into a room, make a gesture or speak aloud and the environment would bend to our will and respond or react as deemed appropriate for that point in time.

Considerable effort has gone into realizing Weiser's vision in terms of developing frameworks, technologies and infrastructures. Proactive computing was put forward as an approach to determine how to program computers to take the initiative to act on people's behalf [43]. The environment has been augmented with various computational resources to provide information and services, when and where desired, with the implicit goal of "assisting everyday life and not overwhelming it" [1]. An assortment of sensors have been experimented with in our homes, hospitals, public buildings, physical environments and even our bodies to detect trends and anomalies, providing a dizzying array of data about our health, movements, changes in the environment and so on. Algorithms have been developed to analyze the data in order for inferences to be made about what actions to take for people. In addition, sensed data is increasingly being used to automate mundane operations and actions that we would have done in our everyday worlds using conventional knobs, buttons and other physical controls. For example, our favorite kind of music or TV show that we like to exercise to will automatically play as we enter a gym. Sensed data is also being used to remind us of things we often forget to do at salient times, such as detecting the absence of milk in the fridge and messaging us to buy a carton when passing the grocery store.

But, as advanced and impressive as these endeavors have been they still do not match up to anything like a world of calm computing. There is an enormous gap between the dream of comfortable, informed and effortless living and the accomplishments of UbiComp research. As pointed out by Greenfield [20] "we simply don't do 'smart' very well yet" because it involves solving very hard artificial intelligence problems that in many ways are more challenging than creating an artificial human

[26]. A fundamental stumbling block has been harnessing the huge variability in what people do, their motives for doing it, when they do it and how they do it. Ethnographic studies of how people manage their lives – ranging from those suffering from Alzheimer's Disease to high-powered professionals – have revealed that the specifics of the context surrounding people's day-to-day living are much more subtle, fluid and idiosyncratic than theories of context have led us to believe [40]. This makes it difficult, if not impossible, to try to implement context in any practical sense and from which to make sensible predictions about what someone is feeling, wanting or needing at a given moment. Hence, while it has been possible to develop a range of simple UbiComp systems that can offer relevant information at opportune moments (e.g., reminding and recommending to us things that are considered useful and important) it is proving to be much more difficult to build truly smart systems that can understand or accurately model people's behaviors, moods and intentions.

The very idea of calm computing has also raised a number of ethical and social concerns. Even if it was possible for Weiser's dream to be fulfilled would we want to live in such a world? In particular, is it desirable to depend on computers to take on our day-to-day decision-making and planning activities? Will our abilities to learn, remember and think for ourselves suffer if we begin to rely increasingly on the environment to do them for us? Furthermore, how do designers decide which activities should be left for humans to control and which are acceptable and valuable for the environment to take over responsibility for?

In this paper I argue that progress in UbiComp research has been hampered by intractable computational and ethical problems and that we need to begin taking stock of both the dream and developments in the field. In particular, we need to rethink the value and role of calm and proactive computing as main driving forces. It is without question that Weiser's enormous legacy will (and should) continue to have an impact on UbiComp developments. However, sufficient time has passed since his untimely death and it should be possible now for researchers to take a critical stance. As part of this exercise, I propose that the field needs to broaden its scope, setting and addressing other goals that are more attainable and down-to-earth. New agendas need also to be outlined that can guide, stimulate and challenge UbiComp (and other) researchers and developers, building upon the growing body of research in the field.

To this end, I propose one such alternative agenda which focuses on designing UbiComp technologies for engaging user experiences. It argues for a significant shift from proactive computing to proactive people; where UbiComp technologies are designed not to do things for people but to engage them more actively in what they currently do. Rather than calm living it promotes engaged living, where technology is designed to enable people to do what they want, need or never even considered before by acting in and upon the environment. Instead of embedding pervasive computing

everywhere in the environment it considers how UbiComp technologies can be created as ensembles or ecologies of resources, that can be mobile and/or fixed, to serve specific purposes and be situated in particular places. Furthermore, it argues that people rather than computers should take the initiative to be constructive, creative and, ultimately, in control of their interactions with the world – in novel and extensive ways.

While this agenda might appear to be a regressive step and even an anathema to some ardent followers of Weiser's vision, I argue that it (and other agendas) will turn out to be more beneficial for society than persisting with following an unrealistic goal. Current technological developments together with emerging findings from user studies, showing how human activities have been positively extended by 'bounded' (as opposed to pervasive) technologies, suggest that much can be gained from reconceptualizing UbiComp in terms of designing user experiences that creatively, excitedly, and constructively extend what people currently do. This does not mean that the main tenet of Weiser's vision be discarded (i.e., computers appearing when needed and disappearing when not) but rather we begin to entertain other possibilities – besides calmness – for steering UbiComp research. Examples include extending and supporting personal, cognitive and social processes such as habit-changing, problem-solving, creating, analyzing, learning or performing a skill. Ultimately, research and development should be driven by a better understanding of human activity rather than

what has tended to happen, namely, "daring to intervene, clumsily, in situations that already work reasonably well" [20, p231].

In the remainder of this paper I offer a constructive critique of Weiser's vision and the subsequent research that has followed in its footsteps. I then outline an alternative agenda for UbiComp, highlighting pertinent questions, concerns and illustrative examples of how it can be achieved.

2 Weiser's Vision Revisited and Early Research

To illustrate how his early vision of ubiquitous computing could work, Weiser [47] presented a detailed scenario about a day in the life of Sal, an executive single mother. The scenario describes what Sal gets up to, as she moves from her domestic world to her work place, during which she is perpetually informed of the goings on of her family, neighbors, fellow citizens and work colleagues. With this knowledge she is able to keep up-to-date, avoid obstacles, make the most of her time and conduct her work – all in smooth and effective ways. The scenario emphasizes coziness, comfort and effortlessness:

"Sal awakens: she smells coffee. A few minutes ago her alarm clock, alerted by her restless rolling before waking, had quietly asked "coffee?", and she had mumbled "yes." "Yes" and "no" are the only words it knows.

Sal looks out her windows at her neighborhood. Sunlight and a fence are visible

through one, but through others she sees electronic trails that have been kept for her of neighbors' coming and going during the early morning. Privacy conventions and practical data rates prevent displaying video footage, but time markers electronic tracks on the neighborhood map let Sal feel cozy in her street."

In this small excerpt we see how the world evolves around Sal's assumed needs, where computers, cameras and sensors are embedded into her world to make her life super efficient, smooth and calm. It is as if she glides through life, where everything is done or laid out for her and whenever there is potential for frustration, such as a traffic jam or parking problem, the invisible computers come to her rescue and gently inform her of what to do and where to go. It is worth drawing an analogy here with the world of the landed aristocracy in Victorian England who's day-to-day live was supported by a raft of servants that were deemed to be invisible to them. This scenario also highlights the ethical issues that such an informed world needs to address, namely the importance of establishing appropriate levels of privacy that are considered acceptable by a community (e.g., having abstract digital trails rather than video footage to ensure anonymity).

The core topics raised in Weiser's seminal papers have motivated much subsequent UbiComp research. Most prominent themes are context-aware computing, ambient/ubiquitous intelligence and recording/tracking and monitoring. (N.B. It should be noted that these are not mutually exclusive but overlap in the aims and methods used.)

2.1 Context-Aware Computing

Context-aware computing focuses on detecting, identifying and locating people's movements, routines or actions with a view to using this information to provide

relevant information that may augment or assist a person or persons. Many projects have been conducted under this heading to the extent that it has been noted that ubiquitous computing is sometimes called context-aware computing [12]. In a nutshell, context is viewed as something that can be sensed and measured using location, time, person, activity type and other dimensions. An example of an early context-sensitive application was comMotion that used location information and a speech output system to inform people when they were driving or cycling past a store to buy the groceries they needed [30].

A motivation behind much context-aware computing is to find ways of compensating for limitations in human cognition, e.g., attention, memory, learning, comprehension, and decision-making, through the use of sensor-based and computational tools. For example, augmented cognition – originating in military research – seeks to develop methods "to open bottlenecks and address the biases and deficits in human cognition" by continually sensing the ongoing context and inferring what strategies to employ to help people in their tasks [5].

Key questions in context-aware computing concern what to sense, what form and

what kind of information to represent to augment ongoing activities. A number of location and tagging technologies have been developed, such as RFID, satellite, GPS and ultrasonics, to enable certain categories of information to be tracked and detected. Many of these, however, have been beset with detection and precision limitations, sometimes resulting in unreliable and inaccurate data. Recent advances in cognitive radio technology that is software defined (SDR), promises to be more powerful; wireless systems will be able to locate and link to locally unused radio frequency, based on the ability to sense and remember various factors, such as human behavior, making them more dependable and more aware of their surroundings [4]. The advocates of this new technology portray its potential for highly complex settings, such as combat war zones to help commanders from different friendly forces stay appraised of the latest situation, through voice, data and video links, thereby reducing collateral damage [4].

While newer technological developments may enable more accurate data to be detected and collected it is questionable as to how effectively it can be used. It still involves Herculean efforts to understand, interpret and act upon in real-time and in meaningful ways. Context-aware systems that attempt to guide a person through certain activities require models of human behavior and intentionality that are based on rationality and predictability [40]. However, as already mentioned, people often behave in unpredictable and subtle ways in their day-to-day contexts. Therefore, it is likely that context-aware systems will only ever be successful in highly constrained settings.

2.2 Ambient and Ubiquitous Intelligence

Another dominant theme that has emerged in the field of UbiComp is ubiquitous or ambient intelligence, i.e., computational intelligence that is part of both the physical and the digital worlds. This approach follows on from work in artificial intelligence. The phrase ‘right place/right time/right means’ has been sloganized with visions of smart worlds and smart things, embedded with intelligence, that will predict people’s needs and react accordingly [25]. Instead of reaching for the remote to change the TV

channel the smart entertainment system will do it for us, instead of browsing the web the smart internet will find the information we need and so on. Just as it is becoming increasingly common place for supermarkets to automatically open their doors as we walk towards them, toilets to flush when we stand up and taps to release water as we wave our hands under them it is envisioned that information will appear on our TVs, watches, walls, and other displays as and when needed (e.g., children will be alerted of dangers and tourists will be informed of points of interest when walking through an unfamiliar city).

However, similar to context-aware computing, ambient intelligence is proving to be a hard nut to crack. While there have been significant advances in computer vision,

speech recognition and gesture-based detection, the reality of multimodal interfaces – that can predict and deliver with accuracy and sensitivity what is assumed people want or need – is a long way off. One of the most well known attempts at implementing ambient intelligence was IBM's BlueEyes project, that sought to develop computers that could "see" and "feel" like humans. Sensing technology was used to identify a person's actions and to extract key information that was then analyzed to determine the person's physical, emotional, or informational state. This was intended to be used to help make people "more productive by performing expected actions or by providing expected information." The success of the BlueEyes project, however, was limited; an example of an achievement that is posted on its website is of a television that would turn itself on when a person in the room made eye contact with it. To turn it off, the person could 'tell' it to switch off.

Such meager accomplishments in both context-aware computing and ambient intelligence reflect just how difficult it can be to get a machine to behave like a human. But it is essential that such systems be accurate for them to be accepted by humans in their everyday context. Reading, interpreting and acting upon people's moods, intentions, desires, etc, at any given moment in an appropriate way is a highly developed human skill that when humans get it wrong can lead to misunderstanding. When a ubiquitous computing system gets it wrong – which is likely to be considerably more frequent – it is likely to be more frustrating and we are likely to be less forgiving. For example, when the system decides to switch on the TV because we happen momentarily to stare into space while reading a book, it is likely to be unnerving and extremely annoying, especially if 'it' persistently gets it wrong.

2.3 Recording, Tracking and Monitoring

The push towards developing assistive applications through sensing and alerting has been most marked for vulnerable people; a number of UbiComp systems have been built to constantly check up on the elderly, the physically and mentally disabled [34]. The movements, habits, health and mishaps of such people are recorded, tracked and presented via remote monitors to the families, carers and other people responsible for them, who can then use the information to make decisions about whether to intervene or administer alternative forms of medical care or help. In particular, there has been a move towards developing ubiquitous computing systems to aid elderly people, who need to be cared for, by helping them take their medicines regularly, checking up on their physical health, monitoring their whereabouts and detecting when they have fallen over [e.g., 13].

A number of assisted living applications and services has also been developed to help people with loss of vision or deteriorating memory to be more independent in their lives. For example, Cyber Crumbs was designed to help people with progressive vision loss find their way around a building using a reader badge system that reads out

directions and warns of obstacles, such as fire hydrants [39]. Cook's Collage was developed as an aid for people with memory loss. It replays a series of digital still images in a comic strip reel format depicting people's cooking actions in situ, intended to help them remember if they have forgotten a step (e.g., adding a particular ingredient) after being distracted [45].

A reason for there being so much interest in helping the less able in UbiComp is that explicit needs and benefits can be readily identified for these user groups. Moreover, there is an assumption that pervasive technologies offer more flexibility and scope for providing solutions compared with other computing technologies since they can sense, monitor and detect people's movements, bodily functions, etc., in ways not possible before. There is a danger, however, that such techniques may probe too far into the lives of less able people resulting in – albeit unintentionally – 'extreme' forms of recording, tracking and monitoring that these people may have no control over. For example, consider the extent to which a group of researchers went to in order to help with the care of old people in a residential care home [6]. A variety of monitoring devices were installed in the home, including badges on the patients and the caregivers and switches on the room doors that detected when they were open or closed. Load sensors were also used to measure and monitor weight changes of people while in their beds; the primary aim was to track trends in weight gain or loss over time. But the sensors could also be used to infer how well someone was sleeping. If significant movement was detected during the night this could enable a caregiver to see whether the person was having trouble sleeping (and if there was a huge increase in weight this could be inferred as someone else getting in or on the bed).

Such panopticon developments elicit a knee-jerk reaction of horror in us. While the motives behind such projects are altruistic they can also be naïve, overlooking how vulnerable people's privacy and self-respect may be being violated. Not surprisingly, there has been enormous concern by the media and other social scientists about the social implications of recording, tracking and re-representing people's movements, conversations, actions and transactions. Inevitably, a focus has been on the negative aspects, namely a person's right to privacy being breached. Is it right to be videoing and sensing people when sleeping, eating, etc., especially when they are not at their best [2]? Is it right to be providing information to other family members about their granny's sleeping habits, especially if it can be inferred from the sensed data that she might have got into bed with another patient, which none of the vested parties might want to share or let the others know about.

While most projects are sensitive to the privacy and ethical problems surrounding the monitoring of people, they are not easy to solve and have ended up overwhelming UbiComp research. Indeed, much of the discussion about the human aspects in the field has been primarily about the trade-offs between security and privacy, convenience and privacy, and informedness and privacy. This focus has often been at the

expense of other human concerns receiving less airing, such as how recording, tracking and re-representing movements and other information can be used to facilitate social and cognitive processes.

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My intention here is not to diminish the importance of awareness, ambience and monitoring to detect and inform people in their everyday lives, together with the ethical and social issues they raise. Rather, my overview of the projects in these areas has revealed how difficult it is to build calm computing systems and yet the attempts have largely dominated the field of UbiComp. Those that have tried have fallen short, resulting in prototype systems that can sometimes appear to be trivial or demeaning. Conversely, there has been less focus on other areas of research that could prove to be easier to achieve and potentially of more benefit to society. The time is ripe for other directions to take center stage in UbiComp. One such avenue promoted here is to consider how humankind's evolved practices of science, learning, health, work and play can be enhanced. This involves thinking about UbiComp not in terms of embedding the environment with all manner of pervasive technologies but instead as bounded ensembles of entities (e.g., tools, surfaces and lenses) that can be mobile, collaborative or remote, through which information, other people and the environment are viewed and interacted with when needed. Importantly, it argues for rethinking the nature of our relationship with the computer.

3 A New Agenda for UbiComp: Engaging User Experiences

I suggest here that it is highly profitable to recast UbiComp research in the context of a central motivation that computers were originally designed for, namely, as tools, devices and systems that can extend and engage people in their activities and pursuits. My reason for proposing this is based on the success of researchers who have started to take this approach. In particular, a number of user studies, exploring how UbiComp technologies are being appropriated, are revealing how the 'excitement of interaction' can be brought back in innovative ways; that is not frustrating and which is quite different from that experienced with desktop applications. For example, various mixed reality, physical-digital spaces and sensor-rich physical environments have been developed to enable people to engage and use multiple dynamic representations in novel ways: in scientific and working practices and in collaborative learning and experimental games. More extensive inquiries and decisions have been enabled in situ, e.g., determining the effects of deforestation in different continents and working out when is the best time to spray or pick grapes in a vineyard.

Recently, world famous computer scientist John Seely Brown put forward his updated vision of UbiComp 1 in a keynote, outlining 'a common sense' model that emphasizes how UbiComp can help to catalyze creativity [41]. He proposed that creating and learning be seen as integral to our work and leisure that are formed through re-creation and appropriation activities. In a similar vein, I argue that it is timely to

switch from a reactive view of people towards a more proactive one. Instead of augmenting the environment to reduce the need for humans to think for themselves about what to do, what to select, etc., and doing it for them, we should consider how UbiComp technologies can be designed to augment the human intellect so that people can perform ever greater feats, extending their ability to learn, make decisions, reason, create, solve complex problems and generate innovative ideas. Weiser's idea that

1 John Seely Brown was a co-author of the paper written by Weiser on calm technology.

technologies be designed to be 'so embedded, so fitting and so natural' that we use them without thinking about them needs to be counter-balanced; we should also be designing them to be exciting, stimulating and even provocative – causing us to reflect upon and think about our interactions with them. While Weiser promoted the advantages of calm computing I advocate the benefits of engaging UbiComp experiences that provoke us to learn, understand and reflect more upon our interactions with technologies and each other.

A central concern of the engaging UbiComp experiences agenda is to fathom out how best to represent and present information that is accessible via different surfaces, devices and tools for the activity at hand. This requires determining how to make intelligible, usable and useful, the recordings of science, medicine, etc., that are streaming from an increasing array of sensors placed throughout the world. It also entails figuring out how to integrate and replay, in meaningful and powerful ways, the masses of digital recordings that are begin gathered and archived such that professionals and researchers can perform new forms of computation and problem-solving, leading to novel insights. In addition, it involves experimenting more with creative and constructive uses of UbiComp technologies and archived digital material that will excite and even make people feel uncomfortable.

In terms of who should benefit, it is useful to think of how UbiComp technologies can be developed not for the Sal's of the world, but for particular domains that can be set up and customized by an individual firm or organization, such as for agriculture production, environmental restoration or retailing. At a smaller scale, it is important to consider how suitable combinations of sensors, mobile devices, shared displays, and computational devices can be assembled by non-UbiComp experts (such as scientists, teachers, doctors) that they can learn, customize and 'mash' (i.e., combine together different components to create a new use). Such toolkits should not need an army of computer scientists to set up and maintain, rather the inhabitants of ubiquitous worlds should be able to take an active part in controlling their set up, evolution and destruction. Their benefits should be clear: enabling quite different forms of information flow (i.e., ways and means of accessing information) and information management (i.e., ways of storing, recording, and re-using information) from older technologies, making it possible for non-UbiCompers to begin to see how to and subsequently develop their

own systems that can make a difference to their worlds. In so doing, there should be an emphasis on providing the means by which to augment and extend existing practices of working, learning and science.

As quoted by Bruner [10] "to assist the development of the powers of the mind is to provide amplification systems to which human beings, equipped with appropriate skills, can link themselves" (p.53). To enable this to happen requires a better understanding of existing human practices, be it learning, working, communicating, etc. Part of this reconceptualization should be to examine the interplay between technologies and their settings in terms of practice and appropriation [15]. "Practices develop around technologies, and technologies are adapted and incorporated into practices." (Dourish, 2001, p. 204). More studies are needed that examine what people do with their current tools and devices in their surrounding environments. In addition, more studies are needed of UbiComp technologies being used in situ or the wild – to help illuminate how people can construct, appropriate and use them [e.g., 16, 22, 23, 29].

With respect to interaction design issues, we need to consider how to represent and present data and information that will enable people to more extensively compute, analyze, integrate, inquire and make decisions; how to design appropriate kinds of interfaces and interaction styles for combinations of devices, displays and tools; and how to provide transparent systems that people can understand sufficiently to know how to control and interact with them. We also need to find ways of enabling professionals and laypeople alike to build, adapt and leverage UbiComp technologies in ways that extend and map onto their activities and identified needs.

A more engaging and bounded approach to UbiComp is beginning to happen but in a scattered way. Three of the most promising areas are described below: (i) playful and learning practices, (ii) scientific practices and (iii) persuasive practices. They show how UbiComp technologies can be developed to extend or change human activities together with the pertinent issues that need to be addressed. Quite different practices are covered, reflecting how the scope of UbiComp can be broad but at the same time targeted at specific users and uses.

3.1 Playful and Learning Practices

One promising approach is to develop small-scale toolkits and sandboxes, comprising interlinked tools, digital representations and physical artifacts that offer the means by which to facilitate creative authoring, designing, learning, thinking and playing. By a sandbox it is not meant the various senses it has been used in computing but more literally as a physical-digital place, kitted out with objects and tangibles to play and interact with. Importantly, these should allow different groups of people to participate in novel activities that will provoke and extend existing repertoires of technology-augmented learning, playing, improvising and creating. An example of a promising UbiComp technology toolkit is PicoCrickets, developed at MIT Media Lab, arising

from the work of Mitch Resnick and his colleagues. The toolkit comprises sensors, motors, lights, microcomputers, and other physical and electrical devices that can be easily programmed and assembled to make them react, interact and communicate, enabling "musical sculptures, interactive jewelry, dancing creatures and other playful inventions" to be created by children and adults alike. An advantage of such light-weight, off-the-shelf tangible toolkits is that they offer many opportunities for different user groups (e.g., educators, consultants) to assemble and appropriate in a range of settings, such as schools, waiting rooms, playgrounds, national parks, and museums. A nagging question, however, is how do the benefits of such UbiComp toolkits and sand boxes compare with those offered by more conventional ones – that are much cheaper and more practical to make? Is it not the case that children can be highly creative and imaginative when given simply a cardboard box to play with? If so, why go to such lengths to provide them with new tools? The debate is redolent of whether it is better for children to read a book or watch a 3D Imax movie. One is not necessarily better than the other: the two provide quite different experiences, triggering different forms of imagination, enjoyment and reflection. Likewise, UbiComp and physical toys can both provoke and stimulate, but promote different kinds of learning and collaboration among children. However, a benefit of UbiComp toolkits over physical artifacts is that they offer new opportunities to combine physical interaction, through manipulation of objects or tools or through physical body postural movement and

location, with new ways of interacting, through digital technology. In particular, they provide different ways of thinking about the world than interacting solely with digital representations or solely with the physical world. In turn, this can encourage or even enhance further exploration, discovery, reflection and collaboration [35].

Examples of projects that have pioneered the design of novel physical-digital spaces to facilitate creativity and reflection include the Hunting of the Snark [32], Ambient Wood [36], RoomQuake [33] Savannah [17], Environmental Detectives [27], Drift Table [19] and Feeding Yoshi [7]. Each of these have experimented with the use of mobile, sensor and fixed technologies in combination with wireless infrastructures to encourage exploration, invention, and out of the box thinking.

The Hunting of the Snark adventure game provoked young children into observing, wondering, understanding, and integrating their fragmented experiences of novel physical-digital spaces that subsequently they reflected upon and shared as a narrative with each other. A combination of sensor-based, tangible, handheld and wireless technologies was used to create the physical-digital spaces, where an imaginary virtual creature was purported to be roaming around in. The children had to work out how to entice the creature to appear in them and then gather evidence about its personality, moods, etc, by walking with it, feeding it and flying with it. Similarly, Savannah was designed as a physical-digital game to encourage the development of

children's conceptual understanding of animal behavior and interactions in an imaginary virtual world. The project used GPS and handheld computers to digitally overlay a school playing field with a virtual plain. Children took on the roles of lions, had to hunt animals in the virtual savannah and capture them to maintain energy levels. After the game, the children reflected on their experiences by interacting with a visualization on a large interactive whiteboard, that showed the trails they made in the Savannah and the sounds and images that they encountered at specific place.

The Ambient Wood project used an assortment of UbiComp technologies to encourage more self-initiation in inquiry and reflective learning. Various wireless and sensor technologies, devices and representational media were combined, designed and choreographed to appear and be used in an 'ambient' woodland. Several handcrafted listening, recording and viewing devices were created to present certain kinds of digital augmentations, such as sounds of biological processes, images of organisms, and video clips of life cycles. Some of these were triggered by the children's exploratory movements, others were collected by the children, while still others were aggregated and represented as composite information visualizations of their exploratory behavior.

RoomQuake was designed to encourage children to practice scientific investigatory practices: an earthquake was simulated in a classroom using a combination of interconnected ambient media, string and physical styrofoam balls. The ambient media provided dynamic readings of the simulated earthquakes, which students then represented as physical models using the physical artifacts. The combination of computer-based simulations and physical-based artifacts enabled the whole class to take part in the measuring, modeling, interpreting, sparking much debate and reflection among the children about the seismic events.

As part of the Equator collaboration, a number of innovative 'seamful games' have been developed. The inherent limitations of ubiquitous technologies have been deliberately exploited to provoke the players into thinking about and acting upon their significance to the ongoing activity. Two examples are Treasure in which players had

to move in and out of a wireless network connectivity to collect and then deposit gold tokens and Feeding Yoshi where the players were required to feed virtual creatures scattered around a city with virtual fruits that popped up on their displays as a result of their location and activity therein.

Evaluations of this emerging genre of physical-digital spaces for learning and playing have been positive, highlighting enhanced understanding and an immense sense of engagement. Children and adults have been able to step back and think about what they are doing when taking part in the game or learning experience, examining the rationale behind their choices when acting out and interacting with the UbiComp-based technologies in the space. However, many of the pioneering projects were technology, resource and researcher intensive. While guidance is now beginning to appear

to help those wanting to design UbiComp-based learning and playing experiences [e.g., 9, 36] we need also to strive towards creating the next generation of physical-digital spaces and toolkits that will be as easy, cheap and popular to construct as Lego kits once were.

3.2 Scientific Practices

Another area where UbiComp has great potential for augmenting human activities is the practice of scientific inquiry and research. Currently, the sciences are going through a major transformation in terms of how they are studied and the computational tools that are used and needed. Microsoft's 2020 Science report – a comprehensive vision of science for the next 14 years written by a group of internationally distinguished scientists – outlines this paradigm shift [31]. It points out how new conceptual and technological tools are needed that scientists from different fields can "understand and learn from each other's solutions, and ultimately for scientists to acquire a set of widely applicable complex problem solving capabilities". These include new programming, computational, analysis and publication tools. There is much scope, too, for utilizing UbiComp technologies to enhance computation thinking, through integrating sensor-based instrumentation in the medical, environmental and chemical sciences. The ability to deliver multiple streams of dynamic data to scientists, however, needs to be matched by powerful interfaces that allow them to manipulate and share them in new ways, from any location whether in the lab or in the field. Areas where there is likely to be obvious benefits to scientists through the integration of UbiComp and computational tools are environmental science and climate change. These involve collaborative visualization of scientific data, mobile access to data and capture of data from sensors deployed in the physical world. Being able to gain a bigger, better and more accurate picture of the environmental processes may help scientists make more accurate predictions and anticipate more effectively natural disasters, such as tsunamis, volcanoes, earthquakes and flooding. However, it may not simply be a case of more is more. New ways of managing the burgeoning datasets needs to be developed, that can be largely automated, but which also allows scientists to have effective windows, lenses etc., into so that they can interpret and make intelligible inferences from them at relevant times.

The 2020 report notes how tomorrow's scientists will need to make sense of the masses of data by becoming more computationally literate – in the sense of knowing how to make inferences from the emerging patterns and anomalies that the new

generation of software analysis tools provide. To this end, a quite different mindset is needed in schools for how science is taught. The design of new learning experiences that utilize UbiComp technologies, both indoors and outdoors, need to be developed to seed in young children the sense of what is involved in practicing new forms of complex, computational science. An example of how this can be achieved is the em-

bedded phenomena approach; scientific phenomena are simulated using UbiComp technologies, for long periods of time, to create opportunities for groups of students to explore 'patient' science [32]. Essentially, this involves the accumulation, analysis and representation of data collected from multiple computational devices over extended periods of observation in the classroom or other sites. In so doing, it allows students to engage in the collaborative practice of scientific investigation that requires hard computational thinking but which is also exciting, creative and authentic. A core challenge, therefore, is to find ways of designing novel science learning experiences that capitalize on the benefits of combining UbiComp and PC technologies that can be used over extended periods.

3.3 Persuasive Practices

The third area where there is much potential for using UbiComp technologies to engage people is as part of self-monitoring and behavioral change programs. While a range of persuasive technologies (e.g., adverts, websites, posters) has already been developed to change people's attitudes and behaviors, based on models of social learning [18], UbiComp technologies provide opportunities for new techniques. Specifically, mobile devices, such as PDAs coupled with on-body sensors, can be designed to enable people to take control and change their habits or lifestyles to be healthier by taking account of and acting upon dynamically updated information provided by them. For example, Intille and his group are exploring how mobile computational tools for assessing behavioral change, based on social psychology models, can be developed to motivate physical activity and healthy eating.

A key question that needs to be addressed is whether UbiComp technologies are more (or less) effective compared with other technologies in changing behavior. A diversity of media-based techniques (e.g., pop-up warning messages, reminders, prompts, personalized messages) has been previously used to draw people's attention to certain kinds of information to change what they do or think at a given point. In terms of helping people give up habits (e.g., smoking, excessive eating) they have had mixed results since people often relapse. It is in the long-term context that UbiComp technologies may prove to be most effective, being able to monitor certain aspects of people's behavior and represent this information at critically weak moments in a cajoling way. A constant but gentle 'nagging' mechanism may also be effective at persuading people to do something they might not have otherwise done or to not to do something they are tempted to do. For example, a collaborative cell phone application integrated with a pedometer was used to encourage cliques of teenage girls to monitor their levels of exercise and learn more about nutrition in the context of their everyday activities [44]. The software was designed to present the monitored process (e.g., walking) in a way that made it easy for the girls to compute and make inferences of how well they were doing in terms of the number of steps taken relative to each other. A preliminary study showed that such a collaborative self-monitoring system was

effective at increasing the girl's awareness of their diet, level of exercise and enabling them to understand the computations involved in burning food during different kinds of exercise. But most significantly, it enabled the girls to share and discuss this information with each other in their private clique, capitalizing on both the persuasive technology and peer pressure.

Incorporating fun into the interface can also be an effective strategy; for example, Nintendo's Pocket Pikachu with pedometer attached was designed to motivate children into being more physically active on a consistent basis. The owner of the digital pet that 'lives' in the device is required to walk, run or jump each day to keep it alive. If the owner does not exercise for a week the virtual pet becomes unhappy and eventually dies. This can be a powerful means of persuasion given that children often become emotionally attached to their virtual pets, especially when they start to care for them.

UbiComp technologies can also be used to reduce bad habits through explicitly providing dynamic information that someone would not have been aware of otherwise. In so doing, it can make them actively think about their behavior and modify it accordingly. The WaterBot system was developed using a special monitoring and feedback device to reduce householder's usage of water in their homes – based on the premise that many people are simply unaware of how wasteful they are [3]. A sensor-based system was developed that provided positive auditory messages and chimes when the tap was turned off. A central idea was to encourage members of the household to talk to one another about their relative levels of water usage provided by the display and to try to out do one another in the amount of water used.

But to what extent do UbiComp technologies, designed for persuasive uses, differ from the other forms of monitoring that were critiqued earlier in the paper? A main difference is that there is more active involvement of those being monitored in attaining their desired behavior change compared with those who were being monitored and assisted in care homes. The objective is to enable people, themselves, to engage with the collected information, by monitoring, understanding, interpreting and acting upon it – and not the environment or others to act upon their behalf. Much of the research to date in UbiComp and healthcare has focussed on automated bio-monitoring of physiological processes, such as EEGs and heart rate, which others, i.e., specialists, examine and use to monitor their patient's health. In contrast, persuasive technologies are intended to provide dynamic information about a behavioral process that will encourage people from doing or not doing something, by being alerted and/or made aware of the consequences of what they are about to do. Moreover, designing a device to be solely in the control of the users (and their social group) enables them to be the owners of the collected data. This circumvents the need to be centrally concerned with privacy issues, allowing the focus of the research to be more

oriented towards considering how best to design dynamically updated information to support cognitive and social change. A challenge, however, in this area is for long term studies to be conducted that can convincingly show that it is the perpetual and time-sensitive nature of the sensed data and the type of feedback provided that contributes to behavioral modification.

4 Conclusions

Many of the research projects that have followed in the footsteps of Weiser's vision of calm computing have been disappointing; their achievements being limited by the extent to which they have been able to program computers to act on behalf of humans. Just as 'strong' AI failed to achieve its goals – where it was assumed that "the computer is not merely a tool in the study of the mind; rather, the appropriately programmed computer really is a mind" [41], it appears that 'strong' UbiComp is suffering from the same fate. And just as 'weak' AI 2 revived AI's fortunes, so, too, can 'weak' UbiComp bring success to the field. This will involve pursuing more practical goals and addressing less ambitious challenges; where ensembles of technologies are designed for specific activities to be used by people in bounded locations. To make this happen, however, requires moving from a mindset that wants to make the environment smart and proactive to one that enables people, themselves, to be smarter and proactive in their everyday and working practices. Three areas of research were suggested as to how this could be achieved; but, equally, there are others where there is much potential for enhancing and extending human activities (e.g., vineyard computing [11], firefighting [24] and sports). As part of the expansion of UbiComp, a wider range of human aspects should be considered, drawing upon alternative theory, guiding frameworks and metaphors [c.f. 8, 15]. To enable other human concerns to become more prominent, however, requires the hefty weight of privacy and other related ethical issues on UbiComp's shoulders to be lessoned.

The 'excitement of interaction' that Weiser suggested forsaking in the pursuit of a vision of calm living should be embraced again, enabling users, designers and researchers to participate in the creation of a new generation of user experiences that go beyond what is currently possible with our existing bricolage of tools and media. We should be provoking people in their scientific, learning, analytic, creative, playing and personal activities and pursuit. Finally, while we have been privileged to have had such a great visionary, whose legacy has done so much to help shape the field, it is timely for a new set of ideas, challenges and goals to come to the fore and open up the field.

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The Lucent Web site is built hierarchically, in the sense that pages deeper in the directory tree represent more detailed information than those at shallower levels. At its busiest, there can be as many as 300 people browsing www.lucent.com; while during the pre-dawn hours there can be as few as 5 simultaneous visitors. Our sonification is designed to convey qualitative information about site usage, answering questions like:

Overall, is the site busy or quiet?

What proportion of the visitors are delving for specific information deep within the site, as compared to those visitors who are “just passing through,” glancing briefly at the home page and then moving on?

How are users distributed across the various content areas of the site?

Which portions of the site are visited together? What kinds of patterns do we find in user behavior?

We think of this sonification as one possible “background” information stream that can inform content providers, Web designers and even the visitors themselves.

2.1.1. Sonification design

Our audio display makes use of the hierarchical structure of the content offered by www.lucent.com. First, a unique pitch was used to identify each of five high-level subdomains within the site: /micro, representing Lucent’s microelectronics design and manufacturing business (now Agere Systems); /enterprise, for the enterprise systems and software business (now Avaya Communications); /minds, a corporate introduction to Bell Labs research; /press, a collection of press releases and investor information; and /search, the local search engine for the site.

The total number of visitors accessing any information from a subdomain affects the loudness and tonal balance of a low-register

drone at the associated pitch. Visitors requesting content deeper in the site are represented by higher-pitched pulsing tones (separated by one or two octaves from the base pitch for the subdomain): the faster the pulse, the more people are accessing that area, and the greater the proportion of high-register sounds, the more detailed the content. By assigning well-separated pitches to each subdomain, shifts in activity both within and between the areas can be heard. In Table 1 we present a simple mapping of data collected by the Lucent Web server to a continuously time-varying vector of usage statistics. In the category of Overall browsing, we count any visitor accessing content pages (HTML, PostScript or PDF) from the indicated subdomain. A Mid-Level access is a request for content two or more directories down. Simple examples are /micro/K56flex/index.html (information on a brand of 56K modem) and /press/0101/010118.nsb.html (a press release for January 18, 2001). The final category, Deep browsing, refers to pages that are four or more directories down in the tree. One example is a paper from the April/June 2000 issue of the Bell Labs Technical Journal, located at /minds/techjournal/apr-jun2000/pdf/paper02.pdf.

Then, the resulting 15 values in Table 1, A1–E3, were mapped to sound as follows:

Overall activity Measured by A1–E1, voiced with a low-register drone. The aggregate number of visitors accessing information within each of the five areas modulates the loudness of each of the five pitches.

	/micro	/enterprise	/minds	/press	/search
Overall	A1	B1	C1	D1	E1
Mid-Level	A2	B2	C2	D2	E2
Deep	A3	B3	C3	D3	E3

Table 1: Mapping used for Web site traffic example. Overall activity records the movements of all users; Mid-Level counts users 2 or 3 directories into the site; Deep browsing consists of users 4+ directories down.

Mid-Level browsing Measured by A2–E2 and assigned a rhythmic middle-register tone pulse; pulse loudness and repetition speed rises and the timbral brightness increases as the volume of mid-level browsing increases. There are five independent pulses, each at a different fixed pitch, representing the five content areas.

Deep browsing Measured by A3–E3 and made audible via rhythmic high-register "ting" sounds (plucked steel string samples). Loudness and repetition speed rises as the volume of deep browsing increases. Again, there are five independent "ting" sounds, each at a different fixed pitch, representing the five content areas.

We used pitch groups that were consonant, and for the sounds that incorporated rhythm (A2–E3), the phase and frequency of each pulse in the matrix varies independently, yielding a sound with a changing rhythmic texture but no fixed beat.

The purpose of this sonification is to make interpretable the activities of users on a Web site. Therefore, the stream of hits being processed by a Web server (reduced to include only the HTML, PostScript and PDF documents) needs to be transformed to extract meaningful user-level data. A real-time monitoring tool was developed that maintains a bank of active visits (recording separately the activities of all the people browsing the site at a given time) and updates various statistics with each user request. When cookies or some other authentication mechanism allows us to recognize returning visitors, the monitor will update a more complicated user profile that encapsulates previous browsing patterns. Our traffic sonification as described above takes as input the location of each visitor within a site at a given point in time. When constructing more elaborate sound displays, our design will continue to focus on user activities, drawing more heavily on the statistics culled by the monitoring tool. This emphasis distinguishes our approach from sonification methods that assess Web server performance by making audible statistics relating to server load, HTTP errors, and agent types [?].

2.1.2. Impressions and extensions

We have created three audio examples for the activity on the Lucent site. Our data were captured on November 11, 1999 and we created sonifications of the traffic at 6:00 am, an extremely slow period for the site; noon, a relatively active time; and 2:30 pm, the point at which the site was busiest. The samples are located at our project Web site [6]. Even with this relatively straightforward mapping, one finds compelling patterns. For example, the affinity between the /enterprise subdomain and the /search facility can be heard as the pulses for these areas rise and fall together.3

3 While clearly audible, these shifts can really only be precisely associ-

ated with areas after a certain amount of experience with the mapping.

Also, when comparing moderately active to extremely busy periods, we find that the number of people digging deep into the site is not a fixed fraction of the total number of visitors. That is, the volume of the low-register drones exhibits much more variation than the components for the other two categories of accesses. Each of these effects can be verified by examining the logs, reinforcing the usefulness of our sonification as a tool for constructing hypotheses about site traffic.

As mentioned at the beginning of this section, Web browsers offer a rich set of data about the visitor when requesting data from a server. This display makes use of only the most basic information about a visit, namely the depth of pages accessed. In ongoing work, we are augmenting our sonification with extra features derived both directly from the server data as well as from statistical navigation models [12] fit for the Web site under study. So far, we have found that such extensions are most effective when developed in the context of a particular monitoring application. For example, an extended version of this ambient display can aid system architects of large, Web hosting services understand cache performance and can aid in server provisioning. Another extension will make greater user of our navigation models and can help designers and usability engineers better architect Web sites. We will report on these and other developments through the project Web site [4].

2.2. Chat rooms and bulletin boards

At any given moment, tens of thousands of real-time conversations are taking place across the Internet on public forums, bulletin boards and chat sites. To imagine making these conversations simultaneously audible evokes an image of uproarious babble. And yet, in the aggregate, this massive stream of live communication could exhibit rich thematic structure. Can we find a meaningful way to listen in to so many conversations, rendering them in a way that is comprehensible and not overwhelming?

In some sense, a byproduct of our Web traffic sonification is the creation of a kind of community from the informal gathering of thousands of visitors to a given Web site. Traditionally, informational Web sites like www.lucent.com have provided

us with very little sense of the other people who are requesting data from the server. To attract and retain visitors, however, many commercial sites recognize the potential of the Web to form social as well as informational networks. As a result, Web-based forums, message boards and a variety of chat services are common components of current site designs. While Internet Relay Chat (IRC) has been a widely used standard since the inception of the Internet, the popularization of the Web has resulted in a virtual explosion of chat applications.4 For example, www.yahoo.com (a US-based Web portal) offers hundreds of separate chat rooms attracting tens of thousands of visitors a day. Specialized sites like www.style.com (the homepage for Vogue magazine) or www.audiworld.com (an resource for Audi owners) have also found their message boards to be the most frequently accessed parts of their domains.

To get a sense of the amount of content that is available in these dynamic formats, we examined sites contained in the DMOZ Open Directory [3], an open source listing of over 2 million Web sites compiled and categorized by 33,000 volunteer editors. From the November 20, 2000 image of the directory, we counted 36,681 separate sites offering some kind of chat, bulletin board or other public forum. While we did not examine the activity on all of these sites, the number is staggering. If we include other peer-to-peer communication technologies like instant messaging,5 the amount of dialogue taking place on the Web at any point in time is almost unfathomable. The goal of our second sonification is to make interpretable the thousands of streams of dynamic information being generated on the Web. In so doing, we attempt to characterize a global dialogue, integrating political debates, discussions of current events, and casual exchanges between members of virtual communities.

4 RC was developed by Jarkko Oikarinen in Finland in the late eighties, and was originally intended to work as a better substitute for talk on his bulletin board.

2.2.1. Content monitors and the statistics engine

Our starting point is text. Albeit diverse in style and dynamic in character, the text (or transcript) of these data sources carries their meaning. Therefore, any auditory display consisting only of generated tones would not be able to adequately represent the data without a very complex codebook. The design of our sonifica-

tion then depends heavily on text-to-speech (TTS). As with the traffic example in the previous section, we think of the audio output as another background information stream. The incorporation of spoken components in the sound design poses new challenges, both practical and aesthetic. For example, simply voicing every word taking place in a single chat room can produce too much text to be intelligible when played in real-time and can quickly exhaust the listener. Instead, we build a hierarchical representation of the text streams that relies on statistical processing for content organization and summarization prior to display.

Before considering sonification design, we first had to create specialized software agents that would both discover new chat rooms and message boards, as well as harvest the content posted to these sites. (See Figure 1 for an overview of our system architecture.) Most bulletin boards and some chat applications use standard HTML to store visitor contributions. In many cases, a specific login name is required to gain access to the site. For these situations, we constructed a content agent in Perl, as this language provides us the most convenient platform for managing access details (like cookies). The public chat rooms on sites like chat.yahoo.com can be monitored in this way. For IRC we built a configurable Java client that polls a particular server for active channels. Web sites like www.cnn.com (a popular news portal) and www.financialchat.com (a financial community hosting chat services for day traders) offer several IRC rooms, some of which are tightly moderated.

In addition to collecting content, each monitoring agent also summarizes the chat stream, identifying basic topics and updating statistics about the characteristics of the discussion: What percentage of visitors are contributing? How often to they contribute and at what length? Is the room “on topic,” or are many visitors posting comments on very different subjects? Topics are derived from the chat stream using a variant of generalized sequence mining [7] that incorporates tags for the different parts of speech. While the exact details are beyond the scope of this abstract, a generalized sequence is a string of words possibly separated by a wildcard, “*”. For example, if we let A, B and C denote specific “contentful” words (say, nouns, adjectives and adverbs), then AB C , A B C and A B C are all generalized sequences. The wildcard allows us to identify “Gore * disputes * election” from the sentences

5 AOL alone records tens of millions of people using their instant mes-
saging service each month.
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Figure 1: System architecture overview. A large number of content streams (Chat = chat rooms; BB = Bulletin boards) are gathered by specialized agents that transmit them in a homogenized format to the statistics engine. The statistics engine then distills the streams into a much smaller number of configurable text streams as well as a number of descriptive vectors. The sonification engine then "plays" these text and data streams. The entire systems operates in real-time.

"Vice President Gore filed papers to dispute the presidential elec-
tion," "Aides for Gore indicated that he has every reason to dispute
the election", and "Gore is still deciding whether or not to dispute

the election".

As many posts to chat rooms contain spelling mistakes and incorrect grammar, assigning words to different parts of speech is error-prone. However, unlike most applications of statistical natural language processing, our content monitors update their summaries each time new material is posted and downweight older contributions. Because our sonification renders these sources in real-time, small mistakes have little effect on the power of the overall display to convey the ideas being discussed.

Each of the content monitors are periodically polled by the statistics engine (see Figure 1). This Java-application clusters the different chat rooms and bulletin boards based on their topic and numerical summaries. As the topic in a room changes over time, the statistics engine is constantly updating and reformulating cluster membership. Because a content stream can in fact support a number of simultaneous discussions (the threads of a bulletin board, say), we employ a soft-clustering technique. In our initial work, we have used a mixture-based scheme that determines the number of clusters with an MDL (Minimum Description Length) criterion [9]. Each room is then assigned a probability that it belongs to the different groups. This model also provides for topic summarization at the cluster-level. Next, a stochastic framework was developed to sample representative sentences posted to the chat or bulletin board. When a discussion is extremely unstructured, this selection is essentially random sampling from all the contributions added to the chat since the last polling point. In addition to textual data streams, the statistics engine is also responsible for communicating the various ingredients for the display to our sonification engine, Max/MSP [2] (see Figure 1). We have adopted the Open Sound Control [13] protocol from Center for New Music and Audio Technologies to transfer data between the statistics engine (running on a Macintosh with LinuxPPC) and the sonification engine (running on a Macintosh with OS/9).

2.2.2. Sonification design

As with the previous example (Section 2), our goal is to create a sonification that is both communicative and listenable. Here we face the additional challenge of incorporating verbal content. With TTS annotations, it becomes more difficult to intelligibly convey more than one layer of information through the audio channel. Our design incorporates spatialization, pitch and timbral differentia-

tion, and rhythm to achieve clarity in the presentation of the hierarchically structured data coming from the statistics engine. The auditory display cycles through topic clusters, spending relatively more time on subjects being actively discussed by the largest numbers of people. Each different topic is assigned a different pitch group, reinforcing subject changes when they occur. For each cluster, the statistics engine sends three streams of information to the sonification engine:

Topics A continuously updated list of up to ten "topics" (the most frequently appearing words and phrases – generalized sequences – mined from the multiple chat streams associated with the given cluster; the number of topics is configurable, but ten was chosen based on timing considerations);

Content samples A selection of sample sentences, identified by the statistics engine as typical or representative, in which these topics appear;

Content entropy A vector that represents the changing level of entropy in the source data.

The topics are spoken by the TTS system6 at regular intervals in a pitched monotone, and are panned alternately hard left and hard right in the stereo field, creating a sort of rhythmic "call and response." The sample sentences are panned center, and rendered with limited inflection (as opposed to the pitched monotone of the topics). The tonal, rhythmic and spatial qualities of the topics contrasts sufficiently with the sample sentences to create two distinctly comprehensible streams of verbal information.

The entropy vector controls an algorithmic piano score. When entropy is minimal and the discussion in the chat room or bulletin board is very focused on one subject, chords are played rhythmically in time with the rhythmic recitation of the topics. As entropy increases and the conversations diverge, a Gaussian distribution is used to expand the number, range and dynamics of notes that fall between the chords. With this audio component, one can easily differentiate a well-moderated content source from a more free-form, public chat without distracting from the TTS annotations. The piano score also serves a secondary function as an accompaniment to the vocal foreground, enhancing the compositional balance and overall musicality of the sound design.

2.2.3. Sample sonification and impressions

On our project Web site [5], we have a sample chat room sonifi-

cation that cycles through three topics. In this sound file, we are
6 The built-in MacOS TTS capability controlled by Max/MSP.

listening to the output of only three content monitors. Hence, by design, each topic is confined to a single site. The first portion of this example (ending at 1:47 into the sample) concerns the recent recall of Bridgestone tires and was based on a www.cnn.com chat room. This discussion was heavily moderated and hence the backing piano score frequently reduces to a simple rhythm. For our second topic (from 1:47 to 3:21 of the sample) we recorded chat exchanges on www.financialchat.com one morning when Yahoo's stock opened low. In this example, we hear day traders frantically exchanging predictions about when Yahoo's stock will "bounce." The final topic in this sample (from 3:21 to the end) is again from www.cnn.com and treats a recent strike by the Screen Actor's Guild and the American Federation of Television and Radio Artists. This chat room was much less moderated than the previous CNN chat, and the backing piano score reflects that. Although this example does not make full use of the clustering capabilities of the statistics engine, the essence of our sonification design is clear. The audio display provides an informative and accessible representation of dynamic, textual content. The topic and content sample streams are easy to separate, and when placed in the background, call our attention to important new subjects being discussed on the Web.

2.2.4. Applications and Extensions

Our sonification provides an audible interface to the (now) massive amount of dynamic content available on the Web. Given the preprocessing that takes place in the content monitors and the statistics engine, a simple extension is to provide search-like functionality. A user can register interest in a certain topic and "tune" our display to present only rooms where this subject is being discussed. The necessary ingredients to implement this feature are all currently available in the statistics engine. Similarly, one can easily restrict the sites that are used for the display. When a new subject appears that draws the user's interest, it is also trivial to add a feature that would direct the user's browser to one or more chats associated with the topic. As a final extension, we have pro-

vided the content monitors with a configurable list of Web sites that can be used to help disambiguate elements in the chat stream. For example, the day traders speak in ticker symbols. Providing the content monitor with the URL for the ticker symbol look-up service offered by Yahoo allows the content monitor to weave not only company names but also recent company-related headlines directly into the stream fed to the statistics engine.

While we have focused mainly on chat and bulletin boards, this technology can be applied in other settings. We have begun collaborating with the designers of a natural language interface for Web-based help systems. Here, we give voice to the hundreds of simultaneous conversations taking place between Web site visitors and the automated help system. A similar display can be imagined for other natural language interfaces, including search engines like AskJeeves (www.jeeves.com). In general, the practical applications of this summarization and auditory display tool abound.

3. CONCLUSION AND COMMENTS ON COLLABORATIVE RESEARCH

The two applications outlined in this paper are the first outcomes of a collaboration sponsored by Bell Laboratories and the Brooklyn Academy of Music under the Arts in Multimedia project (AIM). The goal of AIM is to bring together researchers (in this case a statistician) and artists (in this case a sound artist), with the objective of advancing our separate agendas through collaborative projects. Our work together is predicated on the notion that sophistication both in data treatment and aesthetics are crucial to the successful design of audio displays. Thus, in each of our examples, we have endeavored to create a result which communicates information clearly, yet at the same time sounds well composed and appealing. Moving forward, it is our intention to apply these techniques both to practical applications, and also to create a series of artworks. These artworks will use our sonification techniques to establish a series of real-time listening posts, both on the Web and in physical locations. The listening posts will tap in to various points of interest on the Internet, using sound to reveal patterns and trends that would otherwise remain hidden.

In terms of applications, we are exploring the use of sonification to support the design, provisioning and monitoring of communication networks. A network operations center (NOC), for example, routinely receives clues about the health of the system in the

form of text messages generated by routers and switches. An audio display installed inside a NOC can act as an early warning system for approaching bottlenecks as well as aid in troubleshooting. By continued exposure to the sound of a "normally" functioning network, operators will be alerted to system changes that could signal problems.

Art emerges unexpectedly from experimentations with new statistical methods or considerations involving practical applications; and new tools for data analysis and modeling develop in response to artistic concerns. Each of us continues to be surprised by the connections that emerge from rethinking familiar problems in a new context. Through our project, we hope to illustrate both the value of art-technology collaborations as well as their necessity, especially when finding meaning in complex data.

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TAXONOMY AND DEFINITIONS FOR SONIFICATION AND AUDITORY DISPLAY
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ABSTRACT
Sonification is still a relatively young research field and many terms such as sonification, auditory display, auralization, audification have been used without a precise definition. Recent developments such as the introduction of Model-Based Sonification, the establishment of interactive sonification and the increased interest in sonification from arts have raised the need to revisit the definitions in order to move towards a clearer terminology. This paper introduces a new definition for sonification and auditory display that emphasizes the necessary and sufficient conditions for organized sound to be called sonification. It furthermore suggests a taxonomy, and discusses the relation between visualization and sonification. A hierarchy of closed-loop interactions is furthermore introduced. This paper aims to initiate vivid discussion towards the establishment of a deeper theory of sonification and auditory display.
1. INTRODUCTION
Auditory Display is still a young research field whose birth may be perhaps best traced back to the first ICAD conference1 in 1992 organized by Kramer. The resulting proceedings volume "Auditory Display" [1] is still one of the most important books in the field. Since then a vast growth of interest, research, and initiatives in auditory display and sonification has occurred. The potential of sound to support hu-

man activity, communication with technical systems and to explore complex data has been acknowledged [2] and the field has been established and has clearly left its infancy. As in every new scientific field, the initial use of terms lacks coherence and terms are being used with diffuse definitions. As the field matures and new techniques are discovered, old definitions may appear too narrow, or, in light of interdisciplinary applications, too unspecific. This is what motivates the redefinitions in this article.

The shortest accepted definition for sonification is from Barrass and Kramer et al. [2]: "Sonification is the use of non-speech audio to convey information". This definition excludes speech as this was the primary association in the

1see www.icad.org

auditory display of information at that time. The definition is unclear about what is meant by conveyance of information: are real-world interaction sounds sonifications, e.g. of the properties of an object that is being hit? Is a computer necessary for its rendition? As a more specific definition, the definition in [2] continues:

"Sonification is the transformation of data relations into perceived relations in an acoustic signal for the purposes of facilitating communication or interpretation."

It is significant that the emphasis here is put on the purpose of the usage of sound. This automatically distinguishes sonification from music, where the purpose is not on the precise perception of what interactions are done with an instrument or what data caused the sound, but on an underlying artistic level that operates on a different level. Often, the word 'mapping' has been used interchangeably with 'transformation' in the above definition. This, however, suggests a severe limitation of sonification towards just mappings between data and sound – which was perfectly fine at the time of the definition where such a 'Parameter-Mapping Sonification' was the dominating paradigm.

However, the introduction of Model-Based Sonification (MBS) [3, 4] demonstrates methods to explore data by using sound in a way that is very different from a mapping: in Parameter-Mapping Sonification, data values are mapped

to acoustic attributes of a sound (in other words: the data ‘play’ an instrument), whereas in MBS sonification models create and configure dynamic processes that do not make sound at all without external interactions (in other words: the data is used to build an instrument or sound-capable object, while the playing is left to the user). The user excites the sonification model and receives acoustic responses that are determined by the temporal evolution of the model. By doing this, structural information is holistically encoded into the sound signal, and is no longer a mere mapping of data to sound. One can perhaps state that data are mapped to the configurations of sound-capable objects, but not that they are mapped to sound.

Clearly, sonification models implemented according to MBS are very much in line with the original idea that sonifi-

cation allows for the discovery of structures in data through sound. Therefore there is the need to reformulate or adapt the definition for sonification to better include such uses of sound, and beyond that hopefully other possible yet-to-be-discovered linkages between data and sound.

Another challenge for the definition comes from the use of sonification in the arts and music: recently more and more artists incorporate methods from sonification in their work. What implications does this have for the term sonification? Think of scientific visualization vs. art: what is the difference between a painting and a modern visualization? Both are certainly organized colors on a surface, both may have aesthetic qualities, yet they operate on a completely different level: the painting is viewed for different layers of interpretation than the visualization. The visualization is expected to have a precise connection to the underlying data, else it would be useless for the process of interpreting the data. In viewing the painting, however, the focus is set more on whether the observer is being touched by it or what interpretation the painter wants to inspire than what can be learnt about the underlying data. Analogies between sonification and music are close-by.

Although music and sonification are both organized sound, and sonifications can sound like music and vice versa, and certainly sonifications can be ‘heard as’ music as pointed out in [5], there are important differences which are so far not manifest in the definition of sonification.

2. A DEFINITION FOR SONIFICATION

This section introduces a definition for sonification in light of the aforementioned problems. The definition has been refined thanks to many fruitful discussions with colleagues as listed in the acknowledgements and shall be regarded as a new working definition to foster ongoing discussion in the community towards a solid terminology.

Definition: A technique that uses data as input, and generates sound signals (eventually in response to optional additional excitation or triggering) may be called sonification, if and only if

(C1) The sound reflects objective properties or relations in the input data.

(C2) The transformation is systematic. This means that there is a precise definition provided of how the data (and optional interactions) cause the sound to change.

(C3) The sonification is reproducible: given the same data and identical interactions (or triggers) the resulting sound has to be structurally identical.

(C4) The system can intentionally be used with different data, and also be used in repetition with the same data.Data Sonification

Algorithm

systematic

transformation reproducable exchangeability

of data

interactions (optional)

Definition: Sonification

Figure 1: Illustration of the general structure and necessary conditions for sonification. The yellow box depicts besides the sonification elements few other components of auditory displays, see also Sec. 3.

This definition emphasizes important prerequisites for the scientific utility of sonification. It has several partly unexpected implications that are to be explored in the follow-

ing discussion.

2.1. Discussion

2.1.1. General Comments

Sonification Techniques: According to the above definition, the techniques Audification, Earcons, Auditory Icons, Parameter-Mapping Sonification as well as Model-Based Sonification are all covered by the definition – they all represent information/data by using sound in an organized and well-structured way and they are therefore different sonification technique.2 This may first appear unfamiliar in light of the common parlance to see earcons/auditory icons as different from sonification. However, imagine an auditory display for biomedical data that uses auditory icons as sonic events to represent different classes (e.g. auditory icons for benign/malignant tissue). The sonification would then be the superposition or mixture of all the auditory icons chosen for instance according to the class label and organized properly on the time axis. If we sonify a data set consisting only of a single data item we naturally obtain as an extreme case a single auditory icon. The same can be said for earcons. Although sonification originally has the connotation of representing large and complex data sets, it makes sense for the definition to also work for single data points.

Data vs. Information: A distinction between data and information is – as far as the above definition – irrelevant. Think of earcons to represent computer desktop interactions such as “delete file”, “rename folder”. There can be a lexicon of terms (file, folder, link) and actions (delete, rename, etc.), and in practical computer implementations these features would be represented numerically, e.g. object = O1, action = A3. By doing so, the information has been turned into data, and this is generally done if there is more than one signal type to give. Information like for instance a verbal message can always be represented numerically and thus be understood as data. On the other side, raw data

2they are also covered by the definition of sonification as ‘non-speech use of sound to convey information’!

values often carry semantic interpretation: e.g. the outside temperature data value -10◦C (a one-dimensional data set of size 1) – this is cold, and clearly information! Assuming that information is always encoded as data values for its processing we can deal with both in a single definition. How the data are then represented by using sound is another question: whether sonification techniques use a more symbolic or analogic representation according to the analogic-symbolic continuum of Kramer [6] is secondary for the definition.

Mapping as a specific case of sonification: Some articles have used "sonification" to refer specifically to mapping-based sonification, where data features are mapped to acoustic features of sound events or streams. Yet sonification is more generally the representation of data by using sound. There may be times when a clear specification of the sonification technique, e.g. as model-based, audification or parameter-mapping sonification, may be helpful to avoid confusion with the general term of sonification. It makes sense to always use the most specific term possible, that is to use the term Parameter Mapping Sonification, Audification, Model-Based Sonification, etc. to convey exactly what is meant. The term Sonification, however, is, according to the definition, more general which is also supported by many online definitions3. In result we suggest using sonification with the same level of generality as the term visualization is used in visual display.

Sonification as algorithm and sound: Sonification refers to the technique and the process, so basically it refers to the algorithm that is at work between the data, the user and the resulting sound. Often, and with equal right, the resulting sounds are called sonifications. Algorithm means a set of clear rules, independent of whether it is implemented on a computer or any other way.

Sonification as scientific method: According to the definition, sonification is an accurate scientific method which leads to reproducible results, addressing the ear rather then the eye (as visualization does). This does not limit the use of sonifications to data from the sciences, but only states that sonification can be used as a valid instru-

ment to gain insight. The subjectivity in human percep-
3http://en.wikipedia.org/wiki/Sonification,
http://wvvel.csee.wvu.edu/sepscor/sonification/lesson9.html,
http://www.techfak.uni-bielefeld.de/ags/ni/projects/datamining/datason/
datason e.html, http://www.cs.uiowa.edu/ kearney/22c296Fall02/ Critten-
donSpecialty.pdf, to name a few.
tion and interpretation is shared with other perceptualization
techniques that bridge the gap between data and the human
sensory system. Being a scientific method, a prefix like in
"scientific sonification" is not necessary.
Same as some data visualizations may be 'viewed' as
art, sonifications may be heard as 'music'[5], yet this use
differs from the original intent.

2.1.2. Comments to (C1)

(C1) The sound reflects objective properties or
relations in the input data.

Real-world acoustics are typically not a sonification al-
though they often deliver object-property-specific system-
atic sound, since there is no external input data as requested
in C1. For instance, with a bursting bottle, one can identify
what is the data, the model and the sound, but the process
cannot be repeated with the same bottle. However, using
a bottle that fills with rain, hitting it with a spoon once a
minute can be seen as a sonification: The data here is the
amount of rainfall, which is here measured by the fill level,
and the other conditions are also fulfilled. Tuning a guitar
string might also be regarded as a sonification to adjust the
tension of a string4. These examples show that sonifications
are not limited to computer-implementations according to
the definition, which embraces the possibility of other non-
computer-implemented sonifications.

The borders of sonification and real-world acoustics are
fuzzy. It might be discussed how helpful it is to regard or
denote everyday sounds as sonifications.

2.1.3. Comments to (C2)

(C2) The transformation is systematic. This
means that there is a precise definition pro-
vided of how the data (and optional interac-
tions) cause the sound to change.

What exactly do we mean by "precise"? Some sound

generators use noise and thereby random elements so that sound events will per se sound different on each rendering. In Parameter-Mapping Sonifications, the intentional addition of noise (for instance as onset jitter to increase perceptability of events that would otherwise coincide) is often used and makes sense. In order to include such cases randomness is allowed in the definition, yet it is important to declare where and what random elements are used (e.g. by describing the noise distribution). It is also helpful to give a motivation for the use of such random elements. By using too much noise, it is possible to generate useless sonifications in the sense that they garble interpretation of the underlying data. In the same way it is possible to create useless scientific visualizations.

4thanks to the referee for this example!

2.1.4. Comments to (C3)

(C3) The sonification is reproducible: given the same data and identical interactions (or triggers) the resulting sound has to be structurally identical.

The definition claims reproducibility. This may not strictly be achieved for several reasons: the loudspeakers may generate a different sound at different temperatures, other factors such as introduced noise as discussed above may have been added. The use of the term “structurally identical” in the definition aims to weaken the stronger claim of sample-based identity. Sample-based identity is not necessary, yet all possible psychophysical tests should come to identical conclusions.

2.1.5. Comments to (C4)

(C4) The system can intentionally be used with different data, and also be used in repetition with the same data.

Repeatability is essential for a technique to be scientifically valid and useful – otherwise nobody could check the results obtained by using sonification as instrument to gain insight. However, there are some implications by claim-

ing repeatability for what can and cannot be called sonification. It has for instance been suggested that a musician improvising on his instrument produces ‘a sonification of the musician’s emotional state’. With C4, however, “playing a musical instrument” is not a sonification of the performer’s emotional state, since it can not be repeated with the ‘identical’ data. However, the resulting sound may be called a sonification of the interactions with the instrument (regarded here as data), and in fact, music can be heard with the focus to understand the systematic interaction patterns with the instruments.

Some of these conditions have been set as constraints for sonification, e.g. reproducibility in the ‘Listening to the Mind Listening’ concert5, but not been connected to a definition of sonification.

In summary, the given definition provides a set of necessary conditions for systems and methods to be called sonification. The definition is neither exhaustive nor complete; we hope it will serve as the core definition as we as community work towards a complete one.

3. SONIFICATION AND AUDITORY DISPLAY

With the above definition, the term sonification takes the role of a general term to express the method of rendering

5http://www.icad.org/websiteV2.0/Conferences/ICAD2004/concert call.htm

sound in an organized and well-structured way. This is in good analogy with the term visualization which is also the general term under which a variety of specific techniques such as bar charts, scatter plots, graphs, etc. are subsumed. Particularly there is an analogy between scatter plots where graphical symbols (data-mapped color/size...) are organized in space to deliver the visualization, and Parameter-Mapping Sonification, where in a structurally identical way acoustic events (with data-mapped features) are organized in time. It is helpful to have with sonification a term that operates on the same level of generality as visualization.

This raises the question what then do we mean by auditory displays? Interestingly, in the visual realm, the term ‘display’ suggests a necessary but complementary part of the interface chain: the device to generate structured light/images, for instance a CRT or LCD display or a projec-

tor. So in visualization, the term visualization emphasizes the way how data are rendered as an image while the display is necessary for a user to actually see the information. For auditory display, we suggest to include this aspect of conversion of sound signals into audible sound, so that an auditory display encompasses also the technical system used to create sound waves, or more general: all possible transmissions which finally lead to audible perceptions for the user. This could range from loudspeakers over headphones to bone conduction devices. We suggest furthermore that auditory display should also include the user context (user, task, background sound, constraints) and the application context, since these are all quite essential for the design and implementation. Sonification is thereby an integral component within an auditory display system which addresses the actual rendering of sound signals which in turn depend on the data and optional interactions, as illustrated in Fig. 2.

Figure 2: Auditory Displays: systems that employ sonification for structuring sound and furthermore include the transmission chain leading to audible perceptions and the application context.

tion since for instance dialogue systems and speech interfaces may also be regarded as auditory displays since they use sound for communication. While such interfaces are not the primary focus in this research field the terminology suggests their inclusion. On the other hand, Auditory Display may be seen as a subset of the more general term of Auditory Interfaces which do not only include output interfaces (auditory displays, sonification) but also auditory input interfaces which engender bidirectional auditory control and communication between a user and a (in most cases) technical system (e.g. voice control system, query-by humming systems, etc.).

4. HIERARCHY FROM SOUND TO SONIFICATION

So far we have dealt with the necessary conditions surrounding sonification and thus narrowed sonification down to a specific subset of using sound. In this section, we look at sonification in a systemic manner to elucidate its superordinate categories. Figure 3 shows how we suggest to organize the different classes of sound. On the highest level,Map of Sound
Organized Sound
Functional Sounds
Music &
Media Arts Sonification(a)
(b)
Figure 3: Systemic map of sound, showing sonification and its relation to other categories.
sounds are here classified as Organized Sound and unorganized sound. Organized sounds separate from random or otherwise complex structured sounds in the fact that their occurence and structure is shaped by intention. Environmental sounds appear often to be very structured and could thus also be organized sounds, however, if so, any sound would match that category to some extent. It thus may be useful to apply the term to sounds that are intentionally organized – in most cases by the sound/interface developer.
The set of organized sound comprises two large sets that partially overlap: music and functional sounds. Music is

without question a complex structured signal, organized on various levels, from the acoustic signal to its temporal organization in bars, motifs, parts, layers. It is not our purpose to give a definition of music.

The second set is functional sounds. These are organized sounds that serve a certain function or goal [7]. The function is the motivation for their creation and use. To give an example, all signal sounds (such as telephones, doorbells, horns and warning hooters) are functional sounds. Certainly there are intersections with music, as music can serve functional aspects. For instance, trombones and kettle drums have been used to demonstrate kingship and power. A more subtle function is the use of music in supermarkets to enhance the 'shopping mood'. For that reason these sets overlap – the size of the overlap depends on what is regarded as function.

Sonification in the sense of the above definition is certainly a subset of functional sounds. The sounds are rendered to fulfill a certain function, be it communication of information (signals & alarms), the monitoring of processes, or to support better understanding of structure in data under analysis. So is there a difference between functional sounds and sonification at all? The following example makes clear that sonification is really a subset: Recently a new selective acoustic weapon has been used, the mosquito device6, a loudspeaker that produces a HF-sound inaudible to older people, which drives away teenagers hanging around in front of shops. This sound is surely functional, yet it could neither pass as sonification nor as music.

Finally, we discuss whether sonification has an intersection with music&media arts. Obviously there are many examples where data are used to drive aspects of musical performances, e.g. data collected from motion tracking or biosensors attached to a performer. This is, concerning the involved techniques and implementations similar to mapping sonifications. However, a closer look at our proposed definition shows that often the condition for the transformation to be systematic C2 is violated and the exact rules are not made explicit. But without making the relationship explicit, the listener cannot use the sound to understand the underly-

ing data better. In addition, condition C4 may often be violated. If sonification-like techniques are employed to obtain a specific musical or acoustic effect without transparency between the used data and details of the sonification techniques, it might, for the sake of clarity, better be denoted as ‘data-inspired music’, or ‘data-controlled music’ than as sonification. Iannis Xenakis, for instance, did not even want the listener to be aware of the data source nor the rules of sound generation.

6see http://www.compoundsecurity.co.uk/, last seen 2008-01-16

5. CLOSED INTERACTION LOOPS IN AUDITORY DISPLAYS

This section emphasizes the role of interaction in sonification. We propose different terms depending on the scope of the closure of the interaction loop. The motivation for this discussion is that it might be helpful to address how terms such as biofeedback or interactive sonification relate to each other.

We start the discussion with Fig. 4 that depicts closed loop interactions. The sonification module in the upper center playing rendered sonifications to the user. Data sources for sonification enter the box on the left side and the most important parts are (a) World/System: this comprises any system in the world that is connected to the sonification module, e.g. via sensors that measure its state, and (b) Data: these are any data under analysis or represented information to be displayed that are stored separately and accessible by the sonification.World/System
Sonification
Interactive Sonification
Human Activity (supported by sonification)
Auditory Biofeedback
Data
Navigation
Monitoring
No Action

Figure 4: Illustration of Closed-Loop Auditory Systems.

In this setting, Process Monitoring is the least interactive sonification, where data recorded from the world (in real-time) or read from the data repository is continuously used as input for a sonification rendering process. Here, the listener is merely passively listening to the sound with the only active component being his/her focus of attention onto parts of the sound. Certainly, certain changes in the sound might attract attention and force the user to act (e.g. sell stocks, stop a machine, etc...).

A higher degree of active involvement occurs when the user actively changes and adjusts parameters of the sonification module, or interacts otherwise with the sonification system. We denote this case as Interactive Sonification. There is a wide field of possibilities of why and how to do so, and we discuss 3 different prototypical examples:

(a) Triggering: Consider a mapping sonification of a given data set. An essential interaction for the user is to issue the command to render/playback the sonification for a selected dataset. Possibly he/she does this several times in order to attend different parts of the sound signal. This elementary case is an interaction, however, a very basic one.

(b) Parameter Adjustment is done when the user changes parameters, such as what data feature are mapped to acoustic parameters, control ranges, compression factors, etc. Often such adjustments happen separate from the playback so that the changes are made and afterwards the updated sound is rendered. However, interactive real-time control is feasible in many cases and shows a higher degree of interactivity. The user actively explores the data by generating different 'views' of the data [8]. In visualization a similar interactivity is obtained by allowing the user to select axes scalings, etc.

(c) Excitatory Interaction is the third sort of interaction and is structurally similar to the case of triggering. Particularly in Model-Based Sonification [4], usually the data are used to configure a sound-capable virtual object that in turn reacts on excitatory interactions with acoustic responses whereby the user can

explore the data interactively. Excitation puts energy into the dynamic system and thus initiates an audible dynamical system behavior. Beyond a simple triggering, excitatory interactions can be designed to make use of the fine-grained manipulation skills that human hands allow, e.g. by enabling to shake, squeeze, tilt or deform the virtual object, for instance using sensor-equipped physical interfaces to interact with the sonification model. A good example for MBS is Shoogle by Williamson et al. [9], where short text messages in a mobile phone can be overviewed by shaking a mobile phone equipped with accelerometer sensors, resulting in audible responses of the text messages as objects moving virtually inside the phone. Excitatory interactions offer rich and complex interactions for interactive sonification.

The next possibility for a closed loop is by interactions that select or browse data. Since data are chosen, it may best be referred to as Navigation. Navigation can also be regarded as special case of Interactive Sonification, depending on where the data are selected and the borders are here really soft. Navigation usually goes hand in hand with triggering of sonification (explained above).

Auditory Biofeedback can be interpreted as a sonification of measured sensor data. In contrast to the above types, the user's activity is not controlling an otherwise autonomous sonification with independent data, but it produces the input data for the sonification system. The user perceives a sound that depends on his/her own activity. Such systems have applications that range from rehabilitation training to movement training in sports, e.g. to perform

a complex motion sequence (e.g. a tennis serve) so that its sonification is structurally more similar to the sonification of an expert performing the action [10].

The final category is Human Activity, which means that the interaction ranges beyond the sonification system into the world, often driven by the goal to change a world

state in a specific way. In turn, any sensors that pick up the change may lead to changes in the sonification. The difference between the loop types before is that the primary focus is to achieve a goal beyond the sonification system, and not to interact with a closed-loop sonification system. Even without attending the sonification consciously or primarily, the sound can be helpful to reach the goal. For example, imagine the real-world task to fill a thermos bottle with tea. While your primary goal is to get the bottle filled you will receive the 'gluck-gluck' sound with increasing pitch as a by-product of the interaction. If this is consistently useful, you subconsciously adapt your activity to exploit the cues in the sound – but the sound is only periphery for the goal. In a similar sense, sonifications may deliver helpful by-products to actions that change the world state. We regard such interaction add-ons where sonification is a non-obtrusive yet helpful cue for goal attainment as inspiring design direction. Such sonifications might even become subliminal in the sense that users, when asked about the sound, are not even aware of the sound, yet they perform better with sound than without.

6. DISCUSSION AND CONCLUSION

The definitions in this paper are given on the basis of three goals: (i) to anchor sonification as a precise scientific method so that it delivers reproducible results and thus can be used and trusted as instrument to obtain insight into data under analysis. (ii) to offer a generalization which does not limit itself to the special case of mappings from data to sound, but which introduces sonification as general systematic mediator between data and sound, whatever the representation might be. (iii) to balance the definition so that the often-seen pair of terms 'visualization & sonification' are at the same level of generality.

The definition has several implications which have been discussed in Sec. 2. We'd like to emphasize that this effort is being done in hope that the definition inspires a general discussion on the terminology and taxonomy of the research field of auditory display. An online version of the definition is provided at www.sonification.de with the aim to collect comments and examples of sonifications as well as exam-

ples that are agreed not to be sonifications and which help in turn to improve the definition.

In Section 3, we described integral parts for auditory display so that sonification takes a key component as the technical part involving the rendition of sound. Again, the suggested modules are meant as working hypothesis to be discussed at ICAD.

While the given definitions specified terms on a horizontal level, Section 4 proposes a vertical organization of sound in relation to often used terms. The intersections between the different terms and categories have been addressed with examples.

Finally, we have presented in Section 5 an integrative scheme for organizing different classes of auditory closed loops according to the loop closure scope. It proves helpful to clarify classes of interactive sonifications. We think that grouping existing sonifications according to these categories can be helpful to better find alternative approaches for a given task.

The suggested terminology and taxonomy is the result of many discussions and a thorough search for helpful concepts. We suggest it as working definitions to be discussed at the interdisciplinary level of ICAD in hope to contribute towards a maturing of the fields of auditory display and sonification.

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Nomadic Radio: Scaleable and Contextual Notification for Wearable Audio Messaging.
Conference Paper · January 1999
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Nomadic Radio: Scaleable and Contextual Notification for Wearable Audio Messaging
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ABSTRACT
Mobile workers need seamless access to communication and information services on portable devices. However current solutions overwhelm users with intrusive and ambiguous notifications. In this paper, we describe scaleable auditory techniques and a contextual notification model for providing timely information, while minimizing interruptions. User's actions influence local adaptation in the model. These techniques are demonstrated in Nomadic Radio, an audio-only wearable computing platform.
Keywords
Auditory I/O, passive awareness, wearable computing, adaptive interfaces, interruptions, notifications
INTRODUCTION
In today's information-rich environments, people use a number of appliances and portable devices for a variety of tasks in the home, workplace and on the run. Such devices are ubiquitous and each plays a unique functional role in a user's lifestyle. To be effective, these devices need to notify users of changes in their functional state, incoming messages or exceptional conditions. In a typical office environment, the user attends to a plethora of devices with notifications such as calls on telephones, asynchronous messages on pagers, email notification on desktop computers, and reminders on personal organizers or watches. This scenario poses a number of key problems.
Lack of Differentiation in Notification Cues
Every device provides some unique form of notification. In many cases, these are distinct auditory cues. Yet, most cues are generally binary in nature, i.e. they convey only the occurrence of a notification and not its urgency or dynamic state. This prevents users from making timely decisions about received messages without having to shift focus of attention (from the primary task) to interact with the device and access the relevant information.

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Minimal Awareness of the User and Environment

Such notifications occur without any regard to the user’s engagement in her current activity or her focus of attention. This interrupts a conversation or causes an annoying disruption in the user’s task and flow of thoughts. To prevent undue embarrassment in social environments, users typically turn off cell-phones and pagers in meetings or lectures. This prevents the user from getting notification of timely messages and frustrates people trying to get in touch with her.

No Learning from Prior Interactions with User

Such systems typically have no mechanism to adapt their behavior based on the positive or negative actions of the user. Pagers continue to buzz and cell-phones do not stop ringing despite the fact that the user may be in a conversation and ignoring the device for some time.

Lack of Coordinated Notifications

All devices compete for a user’s undivided attention without any coordination and synchronization of their notifications. If two or more notifications occur within a short time of each other, the user gets confused or frustrated. As people start carrying around many such portable devices, frequent and uncoordinated interruptions inhibit their daily tasks and interactions in social environments.

Given these problems, most devices fail to serve their intended purpose of notification or communication, and thus do not operate in an efficient manner for a majority of their life cycle. New users choose not to adopt such technologies, having observed the obvious problems encountered with their usage. In addition, current users tend to turn off the devices in many situations, inhibiting the optimal operation of such personal devices.

Nature of Interruptions in the Workplace

A recent observational study [4] evaluated the effect of

interruptions on the activity of mobile professionals in their workplace. An interruption, defined as an asynchronous and unscheduled interaction, not initiated by the user, results in the recipient discontinuing the current activity. The results revealed several key issues. On average, sub.jects were interrupted over 4 times per hour, for an average duration slightly over 2 minutes. Hence, nearly 10 minutes per hour

was spent on interruptions. Although a majority of the interruptions occurred in a face-to-face setting, 20% were due to telephone calls (no email or pager activity was analyzed in this study). In 64% of the interruptions, the recipient received some benefit from the interaction. This suggests that a blanket approach to prevent interruptions, such as holding all calls at certain times of the day, would prevent beneficial interactions from occurring. However in 41% of the interruptions, the recipients did not resume the work they were doing prior to it. But active use of new communication technologies makes users easily vulnerable to undesirable interruptions.

These interruptions constitute a significant problem for mobile professionals using tools such as pagers, cell-phones and PDAs, by disrupting their time-critical activities. Improved synchronous access using these tools benefits initiators but leaves recipients with little control over the interactions. The study suggests development of improved filtering techniques that are especially light-weight, i.e. don’t require more attention from the user and are less disruptive than the interruption itself. By moving interruptions to asynchronous media, messages can be stored for retrieval and delivery at more appropriate times.

NOMADIC RADIO: WEARABLE AUDIO MESSAGING

Personal messaging and communication, demonstrated in Nomadic Radio, provides a simple and constrained problem domain in which to develop and evaluate a contextual notification model. Messaging requires development of a model that dynamically selects a suitable notification strategy based on message priority, usage level, and environmental context. Such a system must infer the user’s

attention by monitoring her current activities such as interactions with the device and conversations in the room. The user's prior responses to notifications must also be taken into consideration to adapt the notifications over time. In this paper, we will consider techniques for scaleable auditory presentation and an appropriate parameterized approach towards contextual notification.

Several recent projects utilized speech and audio I/O on wearable devices to present information. A prototype augmented audio tour guide [l] played digital audio recordings indexed by the spatial location of visitors in a museum. SpeechWear [11] enabled users to perform data entry and retrieval using speech recognition and synthesis. Audio Aura [10] explored the use of background auditory cues to provide serendipitous information coupled with people's physical location in the workplace. In Nomadic Radio, the user's inferred context rather than actual location is used to decide when and how to deliver scaleable audio notifications. In a recent paper [13], researchers suggest the use of sensors and user modeling to allow wearables to infer when users should be interrupted by incoming messages. They suggest waiting for a break in the conversation to post a message summary on the user's heads-up display. In this paper we describe a primarily non-visual approach to provide timely information to nomadic listeners, based on a variety of contextual cues.

Nomadic Radio is a wearable computing platform that provides a unified audio-only interface to remote services and messages such as email, voice mail, hourly news broadcasts, and personal calendar events. These messages are automatically downloaded to the device throughout the day and users can browse through them using voice commands and tactile input. The system consists of Java-based clients and remote servers (written in C and Perl) that communicate over wireless LAN, and utilize the telephony infrastructure in the Speech Interface group. Simultaneous spatial audio streams are rendered using a HRTF-based Java audio API. Speech I/O is provided via a networked implementation of AT&T Watson Speech API.

To provide a hands-free and unobtrusive interface to a

nomadic user, the system primarily operates as a wearable audio-only device. The SoundBeam Neckset, a research prototype patented by Nortel for use in hands-free telephony, was adapted as the primary wearable platform in Nomadic Radio. It consists of two directional speakers mounted on the user's shoulders, and a directional microphone placed on the chest (see figure 1). Here information and feedback is provided to the user through a combination of auditory cues, spatial audio rendering, and synthetic speech. Integration of a variety of auditory techniques on a wearable device provides hands-free access and navigation as well as lightweight and expressive notification.

An audio-only interface has been incorporated in Nomadic Radio, and a networked infrastructure for unified messaging has been developed for wearable access [12]. The system currently operates on a Libretto 100 mini-portable PC worn by the user. The key issue addressed in this paper is that of handling interruptions to the listener in a manner that reduces disruption, while providing timely notifications for contextually relevant messages.

USAGE AND NOTIFICATION SCENARIO

The following scenario demonstrates the audio interface and presentation of notifications in Nomadic Radio (no voice commands from the user are shown here).

SCALEABLE AUDITORY PRESENTATION

A scaleable presentation is necessary for delivering sufficient information while minimizing interruption to the listener. Messages in Nomadic Radio are scaled dynamically to unfold as seven increasing levels of notification (see figure 3): silence, ambient cues, auditory cues, message summary, preview, full body, and foreground rendering. These are described further below:

Silence for Least Interruption and Conservation

In this mode all auditory cues and speech feedback are turned-off. Messages can be scaled down to silence when the message priority is inferred to be too low for the message to be relevant for playback or awareness to a user,

based on her recent usage of the device and the conversation level. This mode also serves to conserve processing, power and memory resources on a portable device or wearable computer.

Ambient Cues for Peripheral Awareness

In Nomadic Radio, ambient auditory cues are continuously played in the background to provide an awareness of the operational state of the system and ongoing status of messages being downloaded (see figure 4). The sound of flowing water provides an unobtrusive form of ambient awareness that indicates the system is active (silence indicates sleep mode). Such a sound tends to fade into the perceptual background after a short time, so it does not distract the listener. The pitch is increased during file downloads, momentarily foregrounding the ambient sound. A short e-mail message sounds like a splash while a two-minute audio news summary is heard as faster flowing water while being downloaded. This implicitly indicates message size without the need for additional audio cues and prepares the listener to hear (or deactivate) the message before it becomes available. Such peripheral awareness minimizes cognitive overhead of monitoring incoming messages relative to notifications played as distinct auditory cues, which incur a somewhat higher cost of attention on part of the listener.

Related Work in Auditory Awareness

In ARKola [5], an audio/visual simulation of a bottling factory, repetitive streams of sounds allowed people to keep track of activity, rate, and functioning of running machines. Without sounds people often overlooked problems; with auditory cues, problems were indicated by the machine's sound ceasing (often ineffective) or via distinct alert sounds. The various auditory cues (as many as 12 sounds play simultaneously) merged as an auditory texture, allowed people to hear the plant as a complex integrated process. Background sounds were also explored in ShareMon [3], a prototype application that notified users of file sharing activity. Cohen found that pink noise used to indicate %CPU time was considered "obnoxious", even though users understood the, pitch correlation. However,

preliminary reactions to wave sounds were considered positive and even soothing. In Audio Aura [IO], alarm

sounds were eliminated and a number of “harmonically coherent sonic ecologies” were explored, mapping events to auditory, musical or voice-based feedback. Such techniques were used to passively convey the number of email messages received, identity of senders, and abstract representations of group activity.

Auditory Cues for Notification and Identification

In Nomadic Radio, auditory cues are a crucial means for conveying awareness, notification and providing necessary assurances in its non-visual interface. Different types of auditory techniques provide distinct feedback, awareness and message information.

Feedback Cues

Several types of audio cues indicate feedback for a number of operational events in Nomadic Radio:

1. Task completion and confirmations - button pressed, speech understood, connected to servers, finished playing or loaded/deleted messages.
2. Mode transitions - switching categories, going to non-speech or ambient mode.
3. Exceptional conditions - message not found, lost connection with servers, and errors.

Priority Cues for Notification

In a related project, “email glances” [7] were formulated as a stream of short sounds indicating category, sender and content flags (from keywords in the message). In Nomadic Radio, message priority inferred from email content filtering provides distinct auditory cues (assigned by the user) for group, personal, timely, and important messages. In addition, auditory cues such as telephone ringing indicate voice mail, whereas an extracted sound of a station identifier indicates a news summary.

VoiceCues for Identification

VoiceCues represent a novel approach for easy identification of the sender of an email, based on a unique auditory signature of the person. VoiceCues are created by manually extracting a l-2 second audio sample from the

voice messages of callers and associating them with their respective email login. When a new email message arrives, the system queries its database for a related VoiceCue for that person before playing it to the user as a notification, along with the priority cues. The authors have found VoiceCues to be a remarkably effective method for quickly conveying the sender of the message in a very short duration. This technique reduces the need for synthetic speech feedback, which can often be distracting.

Message Summary Generation

A spoken description of an incoming message can present relevant information in a concise manner. Such a description typically utilizes header information in email messages to convey the name of the sender and the subject of the message. In Nomadic Radio, message summaries are generated for all messages, including voice-mail, news and calendar events. The summaries are augmented by additional attributes of the message indicating category, order, priority, and duration. For audio sources, like voice messages and news broadcasts, the system plays the first 2.5 seconds of the audio. This identifies the caller and the urgency of the call, inferred from intonation in the caller's voice or provides a station identifier for news summaries.

Message Previews using Content Summarization

Messages are scaled to allow listeners to quickly preview the contents of an email or voice message. In Nomadic Radio, a preview for text messages extracts the first 100 characters of the message (a default size that can be user defined). This heuristic generally provides sufficient context for the listener to anticipate the overall message theme and urgency. For email messages, redundant headers and previous replies are eliminated from the preview for effective extraction. Use of text summarization techniques, based on tools such as ProSum' developed by British Telecom, would allow more flexible means of scaling message content. Natural language parsing techniques used in ProSum permit a scaleable summary of an arbitrarily large text document.

A preview for an audio source such as a voice message or news broadcast presents a fifth of the message at a gradually increasing playback rate of up to 1.3 times faster than normal. There are a range of techniques for time-compressing speech without modifying the pitch, however twice the playback rate usually makes the audio incomprehensible. A better representation for content summarization requires a structural description of the audio, based on annotated or automatically determined pauses in speech, speaker and topic changes. Such an auditory thumbnail must function similar to its visual counterpart. A preview for a structured voice message would provide pertinent aspects such as name of caller and phone number, whereas a structured news preview would be heard as the hourly headlines.

Full Body: Playing Complete Message Content

This mode plays the entire audio file or reads the full text of the message at the original playback rate. Some parsing of the text is necessary to eliminate redundant header information and format tags. The message is augmented with summary information indicating sender and subject. This message is generally spoken or played in the background of the listener's audio space.

I http://transend.Iabs.bt.com/prosum/on-line/

Foreground Rendering via Spatial Proximity

An important message is played in the foreground of the listening space. The audio source of the message is rapidly moved closer to the listener, allowing it to be heard louder, and played there for 415" of its duration. The message gradually begins to fade away, moving back to its original position and amplitude for the remaining l/S" of the duration. The foregrounding algorithm ensures that the messages are quickly brought into perceptual focus by pulling them to the listener rapidly. However the messages are pushed back slowly to provide an easy fading effect as the next one is heard. As the message moves its spatial direction is maintained so that the listener can retain a focus on the audio source even if another begins to play.

Hence a range of techniques provide scaleable forms of background awareness, auditory notification, spoken

feedback and foreground rendering of incoming messages.

CONTEXTUAL NOTIFICATION

In Nomadic Radio, context dynamically scales the notifications for incoming messages. The primary contextual cues used include: message priority from email filtering, usage level based on time since last user action, and the likelihood of conversation estimated from real-time analysis of the auditory scene. In our experience these parameters provide sufficient context to scale notifications, however data from motion or location sensors can also be integrated in such a model. A linear and scaleable auditory notification model is utilized, based on the notion of estimating costs of interruption and the value of information to be delivered to the user. This approach is similar to recent work [6] on using perceptual costs and a focus of attention model for scaleable graphics rendering.

Message Priority

The priority of incoming messages is explicitly determined via content-based email filtering using CLUES [9], a filtering and prioritization system. CLUES has been integrated into Nomadic Radio to determine the timely nature of messages by finding correlation between a user's calendar, rolodex, to-do list, as well as a record of outgoing messages and phone calls. These rules are integrated with static rules created by the user for prioritizing specific people or message subjects. When a new email message arrives, keywords from its sender and. subject header information are correlated with static and generated filtering rules to assign a priority to the message. Email messages are also prioritized if the user is traveling and meeting others in the same geographic area (via area codes in the rolodex). The current priorities include: group, personal, very important, most important, and timely. Priorities are parameterized by logarithmically scaling all priorities within a range of 0 to 1. Logarithmic scaling ensures that higher priority messages are weighted higher relative to unimportant or uncategorized messages.

Priority (i) = (log (i) / log (Priority Levels Mu))

Usage Level

One problem with using last actions for setting usage levels is that if a user deactivates an annoying message, that action is again time-stamped. Such negative reinforcements continue to increase the usage level and the related notification. Therefore negative actions such as stopping audio playback or deactivating speech are excluded from generating actions for computing the usage.

Likelihood of Conversation

Conversation in the environment can be used to gauge whether the user is in a social context where an interruption is less appropriate. If the system detects the occurrence of more than several speakers over a period of time, that is an indication of a conversational situation. Auditory events are first detected by adaptively thresholding total energy and incorporating constraints on event length and surrounding pauses. The system uses mel-scaled filter-bank coefficients (MFCs) and pitch estimates to discriminate, reasonably well, a variety of speech and non-speech sounds. HMMs (Hidden Markov Models) capture both the temporal characteristics and spectral content of sound events. The techniques for feature extraction and classification of the auditory scene using HMMs are described in a recent workshop paper [2]. The likelihood of speech detected in the environment is computed for each event in a short window of time. In addition, the probabilities are weighted, such that most recent time periods in the window are considered more relevant for computing the overall Speech Level. We are evaluating the classifier's effectiveness by training it with a variety of speakers and background sounds.

Notification Level

A weighted average for all three contextual cues provides level has an inversely proportional relationship with notification i.e. a lower notification must be provided during high conversation.

Presentation Latency

Latency represents the period of time to wait before playing the message to the listener, after a notification cue is delivered. Latency is computed as a function of the

notification level and the maximum window of time (Latency,& that a lowest priority message can be delayed for playback. The default maximum latency is set to 20 seconds, but can be modified by the user.

were increased. Jane was notified of a group message shortly after the voice message, since the system detected higher usage activity. Hence, the system correctly scaled down notifications when Jane did not want to be bothered whereas notifications were scaled up when Jane started to use the system to browse her messages.

EFFECTIVENESS OF THE NOTIFICATION MODEL

The nature of peripheral awareness and unobtrusive notification on a wearable device requires a usage evaluation that must be conducted on an ongoing and long-term basis. However, the predictive effectiveness of the notification model must first be evaluated on a quantitative basis. Hence, all message and notification parameters are captured for such analysis. Lets consider two actual examples of notification computed for email messages with different priorities. Figure 7 shows an auditory cue generated for a group message (low priority).

The timely message (in figure 8) received greater priority and consequently a higher notification level for summary playback. A moderate latency time (approx. 6 secs.) was chosen. However when the user interrupted the notification by a button press, the summary playback was aborted. The user's action reduced overall weights by 5%.

Dynamic Adaptation of the Notification Model

The user can initially set the weights for the notification model to high, medium, or low (interruption). These weight settings were selected by experimenting with notifications over time using an interactive visualization of message parameters. This allowed us to observe the model, modify weights and infer the effect on notification based on different weighting strategies. Pre-defined weights provide an approximate behavior for the model and help bootstrap the system for novice users. The system also allows the user

to dynamically adjust these weights (changing the interruption and notification levels) by their implicit actions while playing or ignoring messages.

The system allows localized positive and negative reinforcement of the weights by monitoring the actions of the user during notifications. As a message arrives, the system plays an auditory cue if its computed notification level is above the necessary threshold for auditory cues. It then uses the computed latency interval to wait before playing the appropriate summary or preview of the message. During that time, the user can request the message be played earlier or abort any further notification for the message via speech or button commands. If aborted, all weights are reduced by a fixed percentage (default is 5%), a negative reinforcement. If the user activates the message (positive reinforcement) within 60 seconds after the notification, the playback scale selected by the user is used to increase all weights. If the message is ignored, no change is made to the weights, but the message remains active for 60 seconds during which the user's actions can continue to influence the weights.

Figure 6 shows a zoomed view of the extended scenario introduced earlier, focusing on Jane's actions that reinforce the model. Jane received several messages and ignored most of the group messages and a recent personal message (the weights remain unchanged). While in the meeting, Jane interrupted a timely message to abort its playback. This reduced the weights for future messages, and the ones with low priority (group message) were not notified to Jane. The voice message from Kathy, her daughter, prompted Jane to reinforce the message by playing it. In this case, the weights

Continuous local reinforcement over time should allow the system to reach a state where it is somewhat stable and robust in converging to the user's preferred notification. Currently the user's actions primarily adjust weights for subsequent messages, however effective reinforcement learning requires a model that generalizes a notification policy that maximizes some long-term measure of reinforcement [8]; this will be the focus of our future work.

PRELIMINARY EVALUATION

Although the authors have been using and relining these techniques during system development, a preliminary 2-day evaluation was conducted with a novice user, who had prior experience with mobile phones and 2-way pagers. The user was able to listen to notifications while attending to tasks in parallel such as reading or typing. He managed to have casual discussions with others while hearing notifications; however he preferred turning off all audio during an important meeting with his advisor. People nearby sometimes found the spoken feedback distracting if heard louder, however that also cued them to wait before interrupting the user. The volume on the device was lowered to minimize any disruption to others and maintain the privacy of messages. The user requested an automatic volume gain that adapted to the environmental noise level. In contrast to speech-only feedback, the user found the unfolding presentation of ambient and auditory cues allowed sufficient time to switch attention to the incoming message. Familiarization with the auditory cues was necessary. He preferred longer and gradual notifications rather than distinct auditory tones. The priority cues were the least useful indicator whereas VoiceCues provided obvious benefit. Knowing the actual priority of a message was less important than simply having it presented in the right manner. The user suggested weaving message priority into the ambient audio (as increased pitch). He found the overall auditory scheme somewhat complex, preferring instead a simple notification consisting of ambient awareness, Voice&es and spoken text.

The user stressed that the ambient audio provided the most benefit while requiring least cognitive effort. He wished to hear ambient audio at all times to remain reassured that the system was still operational. An unintended effect discovered was that a “pulsating” audio stream indicated low battery power on the wearable device. A “pause” button was requested, to hold all messages while participating in a conversation, along with subtle but periodic auditory alerts for unread messages waiting in queue. The user felt that Nomadic Radio provided appropriate awareness and its

expressive qualities justified its use over a pager. A long-term trial with several nomadic users is necessary to further validate these notification techniques.

CONCLUSIONS

We have demonstrated techniques for scaleable auditory presentation and message notification using a variety of contextual cues. The auditory techniques and notification model have been refined based on continuous usage by the authors, however we are currently conducting additional evaluations with several users. Ongoing work explores adaptation of the notification model based on reinforcement from user behavior over time. Our efforts have focused on wearable audio platforms, however these ideas can be readily utilized in consumer devices such as pagers, PDAs and mobile phones to minimize disruptions while providing timely information to users on the move.

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REFERENCES

The Physics of Sound

Sound lies at the very center of speech communication. A sound wave is both the end product of the speech production mechanism and the primary source of raw material used by the listener to recover the speaker's message.

Because of the central role played by sound in speech communication, it is important to have a good understanding of how sound is produced, modified, and measured. The purpose of this chapter will be to review some basic principles underlying the physics of sound, with a particular focus on two ideas that play an especially important role in both speech and hearing: the concept of the spectrum and acoustic filtering. The speech production mechanism is a kind of assembly line that operates by generating some relatively simple sounds consisting of various combinations of buzzes, hisses, and pops, and then filtering those sounds by making a number of fine

adjustments to the tongue, lips, jaw, soft palate, and other articulators. We will also see that a crucial step at the receiving end occurs when the ear breaks this complex sound into its individual frequency components in much the same way that a prism breaks white light into components of different optical frequencies. Before getting into these ideas it is first necessary to cover the basic principles of vibration and sound propagation.

Sound and Vibration

A sound wave is an air pressure disturbance that results from vibration. The vibration can come from a tuning fork, a guitar string, the column of air in an organ pipe, the head (or rim) of a snare drum, steam escaping from a radiator, the reed on a clarinet, the diaphragm of a loudspeaker, the vocal cords, or virtually anything that vibrates in a frequency range that is audible to a listener (roughly 20 to 20,000 cycles per second for humans). The two conditions that are required for the generation of a sound wave are a vibratory disturbance and an elastic medium, the most familiar of which is air. We will begin by describing the characteristics of vibrating objects, and then see what happens when vibratory motion occurs in an elastic medium such as air. We can begin by examining a simple vibrating object such as the one shown in Figure 3-1. If we set this object into vibration by tapping it from the bottom, the bar will begin an upward and downward oscillation until the internal resistance of the bar causes the vibration to cease.

The graph to the right of Figure 3-1 is a visual representation of the upward and downward motion of the bar.

To see how this graph is created, imagine that we use a strobe light to take a series of snapshots of the bar as it vibrates up and down. For each snapshot, we measure the instantaneous displacement of the bar, which is the difference between the position of the bar at the split second that the snapshot is taken and the position of the bar at rest. The rest position of the bar is arbitrarily given a displacement of zero; positive numbers are used for displacements above the rest position, and negative numbers are used for displacements below the rest position. So,

the first snapshot, taken just as the bar is struck, will show an instantaneous displacement of zero; the next snapshot
will show a small positive displacement, the next will show a somewhat larger positive displacement, and so on. The
pattern that is traced out has a very specific shape to it. The type of vibratory motion that is produced by a simple
vibratory system of this kind is called simple harmonic motion or uniform circular motion, and the pattern that is
traced out in the graph is called a sine wave or a sinusoid.
Figure 3-1. A bar is fixed at one and is set into vibration by tapping it from the bottom. Imagine that
a strobe light is used to take a series of snapshots of the bar as it vibrates up and down. At each snapshot the instantaneous displacement of the bar is measured. Instantaneous displacement is the
distance between the rest position of the bar (defined as zero displacement) and its position at any
particular instant in time. Positive numbers signify displacements that are above the rest position, while negative numbers signify displacements that are below the rest position. The vibratory pattern
that is traced out when the sequence of displacements is graphed is called a sinusoid.

Basic Terminology
We are now in a position to define some of the basic terminology that applies to sinusoidal vibration.
periodic: The vibratory pattern in Figure 3-1, and the waveform that is shown in the graph, are examples of
periodic vibration, which simply means that there is a pattern that repeats itself over time.
cycle: Cycle refers to one repetition of the pattern. The instantaneous displacement waveform in Figure 3-1 shows
four cycles, or four repetitions of the pattern.
period: Period is the time required to complete one cycle of vibration. For example, if 20 cycles are completed in 1
second, the period is 1/20th of a second (s), or 0.05 s. For speech applications, the most commonly used unit of
measurement for period is the millisecond (ms):
1 ms = 1/1,000 s = 0.001 s = 10^{-3} s
A somewhat less commonly used unit is the microsecond (μs):
1 μs = 1/1,000,000 s = 0.000001 s = 10^{-6} s
frequency: Frequency is defined as the number of cycles completed in one second. The unit of measurement for

frequency is hertz (Hz), and it is fully synonymous the older and more straightforward term cycles per second
(cps). Conceptually, frequency is simply the rate of vibration. The most crucial function of the auditory system is to
serve as a frequency analyzer – a system that determines how much energy is present at different signal frequencies.
Consequently, frequency is the single most important concept in hearing science. The formula for frequency is:
f = 1/t, where: f = frequency in Hz
t = period in seconds
So, for a period 0.05 s:
f = 1/t = 1/0.05 = 20 Hz
It is important to note that period must be represented in seconds in order to get the answer to come out in cycles per
second, or Hz. If the period is represented in milliseconds, which is very often the case, the period first has to be
converted from milliseconds into seconds by shifting the decimal point three places to the left. For example, for a
period of 10 ms:
f = 1/10 ms = 1/0.01 s = 100 Hz
Similarly, for a period of 100 μs:
f = 1/100 μs = 1/0.0001 s = 10,000 Hz
The period can also be calculated if the frequency is known. Since period and frequency are inversely related, t
= 1/f. So, for a 200 Hz frequency, t = 1/200 = 0.005 s = 5 ms.
Characteristics of Simple Vibratory Systems
Simple vibratory systems of this kind can differ from one another in just three dimensions: frequency,
amplitude, and phase. Figure 3-2 shows examples of signals that differ in frequency. The term amplitude is a bit
different from the other terms that have been discussed thus far, such as force and pressure. As we saw in the last
chapter, terms such as force and pressure have quite specific definitions as various combinations of the basic
dimensions of mass, time, and distance. Amplitude, on the other hand, will be used in this text as a generic term
meaning "how much." How much what? The term amplitude can be used to refer to the magnitude of displacement,
the magnitude of an air pressure disturbance, the magnitude of a force, the magnitude of power, and so on. In the

present context, the term amplitude refers to the magnitude of the displacement pattern. Figure 3-3 shows two displacement waveforms that differ in amplitude. Although the concept of amplitude is as straightforward as the two waveforms shown in the figure suggest, measuring amplitude is not as simple as it might seem. The reason is that the instantaneous amplitude of the waveform (in this case, the displacement of the object at a particular split second in time) is constantly changing. There are many ways to measure amplitude, but a very simple method called peak-to-peak amplitude will serve our purposes well enough. Peak-to-peak amplitude is simply the difference in amplitude between the maximum positive and maximum negative peaks in the signal. For example, the bottom panel in Figure 3-3 has a peak-to-peak amplitude of 10 cm, and the top panel has a peak-to-peak amplitude of 20 cm. Figure 3-4 shows several signals that are identical in frequency and amplitude, but differ from one another in phase. The waveform labeled 0 o phase would be produced if the bar were set into vibration by tapping it from the bottom. The waveform labeled 180 o phase would be produced if the bar were set into vibration by tapping it from the top, so that the initial movement of the bar was downward rather than upward. The waveforms labeled 90 o phase

and 270 o phase would be produced if the bar were set into vibration by pulling the bar to maximum displacement
and letting go -- beginning at maximum positive displacement for 90 o phase, and beginning at maximum negative
displacement for 270 o phase. So, the various vibratory patterns shown in Figure 3-4 are identical except with respect
to phase; that is, they begin at different points in the vibratory cycle. As can be seen in Figure 3-5, the system for
representing phase in degrees treats one cycle of the waveform as a circle; that is, one cycle equals 360 o. For
example, a waveform that begins at zero displacement and shows its initial movement upward has a phase of 0 o, a
waveform that begins at maximum positive displacement and shows its initial movement downward has a phase of
90 o, and so on.

Figure 3-2. Two vibratory patterns that differ in frequency. The panel on top is higher in frequency
than the panel on bottom.

Figure 3-3. Two vibratory patterns that differ in amplitude. The panel on top is higher in amplitude than the
panel on bottom.

Figure 3-4. Four vibratory patterns that differ in phase. Shown above are vibratory patterns with phases of 0 0, 90 0,
180 0, and 270 0.
The Physics of Sound 5
Springs and Masses
We have noted that objects can vibrate at different frequencies, but so far have not discussed the physical
characteristics that are responsible for variations in frequency. There are many factors that affect the natural
vibrating frequency of an object, but among the most important are the mass and stiffness of the object. The effects
of mass and stiffness on natural vibrating frequency can be illustrated with the simple spring-and-mass systems
shown in Figure 3-6. In the pair of spring-and-mass systems to the left, the masses are identical but one spring is
stiffer than the other. If these two spring-and-mass systems are set into vibration, the system with the stiffer spring
will vibrate at a higher frequency than the system with the looser spring. This effect is similar to the changes in
Time ->
Instantaneous Amplitude
0
90
180
270
0/360
Figure 3-5. The system for representing phase treats one cycle of the vibratory pattern as a circle, consisting of 360 0
. A pattern that begins at zero amplitude heading toward positive values (i.e., heading upward) is designated 0 0 phase; a waveform that begins at maximum positive displacement and shows
its initial movement downward has a phase of 90 o
; a waveform that begins at zero and heads
downward has a phase of 180 o; and a waveform that begins at maximum negative displacement and
shows its initial movement upward has a phase of 270 o. . The four phase angles that are shown above
are just examples. An infinite variety of phase angles are possible.
Figure 3-6. A spring and mass system whose natural vibrating frequency is controlled by two parameters: (1) the stiffness of the spring (the stiffer the spring the higher the natural vibrating

frequency), and (2) the mass of the material that is suspended from the spring (the greater the mass, the
lower the natural vibrating frequency).

frequency that occur when a guitarist turns the tuning key clockwise or counterclockwise to tune a guitar string by
altering its stiffness.1
The spring-and-mass systems to the right have identical springs but different masses. When these systems are
set into vibration, the system with the greater mass will show a lower natural vibrating frequency. The reason is that
the larger mass shows greater inertia and, consequently, shows greater opposition to changes in direction. Anyone
who has tried to push a car out of mud or snow by rocking it back and forth knows that this is much easier with a
light car than a heavy car. The reason is that the more massive car shows greater opposition to changes in direction.
In summary, the natural vibrating frequency of a spring-and-mass system is controlled by mass and stiffness.
Frequency is directly proportional to stiffness (S↑F↑) and inversely proportional to mass (M↑F↓). It is important to
recognize that these rules apply to all objects, and not just simple spring-and-mass systems. For example, we will
see that the frequency of vibration of the vocal folds is controlled to a very large extent by muscular forces that act
to alter the mass and stiffness of the folds. We will also see that the frequency analysis that is carried out by the
inner ear depends to a large extent on a tuned membrane whose stiffness varies systematically from one end of the
cochlea to the other.

Sound Propagation

As was mentioned at the beginning of this chapter, the generation of a sound wave requires not only vibration,
but also an elastic medium in which the disturbance created by that vibration can be transmitted (see Box 3-1 [bell
jar experiment described in Patrick's science book - not yet written]). To say that air is an elastic medium means that
air, like all other matter, tends to return to its original shape after it is deformed through the application of a force.

The prototypical example of an object that exhibits this kind of restoring force is a spring. To understand the
mechanism underlying sound propagation, it is useful to think of air as consisting of collection of particles that are
connected to one another by springs, with the springs representing the restoring forces associated with the elasticity
of the medium. Air pressure is related to particle density. When a volume of air is undisturbed, the individual
particles of air distribute themselves more-or-less evenly, and the elastic forces are at their resting state. A volume of
air that is in this undisturbed state it is said to be at atmospheric pressure. For our purposes, atmospheric pressure
can be defined in terms of two interrelated conditions: (1) the air molecules are approximately evenly spaced, and
(2) the elastic forces, represented by the interconnecting springs, are neither compressed nor stretched beyond their
resting state. When a vibratory disturbance causes the air particles to crowd together (i.e., producing an increase in
particle density), air pressure is higher than atmospheric, and the elastic forces are in a compressed state.
Conversely, when particle spacing is relatively large, air pressure is lower than atmospheric.
1The example of tuning a guitar string is imperfect since the mass of the vibrating portion of the string decreases slightly as the string is
tightened. This occurs because a portion of the string is wound onto the tuning key as it is tightened.
a b c d e f g h i
a b c d e f g h i
a b c d e f g h i
a b c d e f g h i
a b c d e f g h i
a b c d e f g h i
a b c d e f g h i
a b c d e f g h i
a b c d e f g h i
a b c d e f g h i
TIME
Figure 3-7. Shown above is a highly schematic illustration of the chain reaction that
results in the propagation of a sound wave (modeled after Denes and Pinson, 1963).

When a vibrating object is placed in an elastic medium, an air pressure disturbance is created through a chain reaction similar to that illustrated in Figure 3-7. As the vibrating object (a tuning fork in this case) moves to the right, particle a, which is immediately adjacent to the tuning fork, is displaced to the right. The elastic force generated between particles a and b (not shown in the figure) has the effect a split second later of displacing particle b to the right. This disturbance will eventually reach particles c, d, e, and so on, and in each case the particles will be momentarily crowded together. This crowding effect is called compression or condensation, and it is characterized by dense particle spacing and, consequently, air pressure that is slightly higher than atmospheric pressure. The propagation of the disturbance is analogous to the chain reaction that occurs when an arrangement of dominos is toppled over. Figure 3-7 also shows that at some close distance to the left of a point of compression, particle spacing will be greater than average, and the elastic forces will be in a stretched state. This effect is called rarefaction, and it is characterized by relatively wide particle spacing and, consequently, air pressure that is slightly lower than atmospheric pressure.

The compression wave, along with the rarefaction wave that immediately follows it, will be propagated outward at the speed of sound. The speed of sound varies depending on the average elasticity and density of the medium in which the sound is propagated, but a good working figure for air is about 35,000 centimeters per second, or approximately 783 miles per hour. Although Figure 3-7 gives a reasonably good idea of how sound propagation works, it is misleading in two respects. First, the scale is inaccurate to an absurd degree: a single cubic inch of air contains approximately 400 billion molecules, and not the handful of particles shown in the figure. Consequently, the compression and rarefaction effects are statistical rather than strictly deterministic as shown in Figure 3-7.

Second, although Figure 3-7 makes it appear that the air pressure disturbance is propagated in a simple straight line

from the vibrating object, it actually travels in all directions from the source. This idea is captured somewhat better
in Figure 3-8, which shows sound propagation in two of the three dimensions in which the disturbance will be
transmitted. The figure shows rod and piston connected to a wheel spinning at a constant speed. Connected to the
piston is a balloon that expands and contracts as the piston moves in and out of the cylinder. As the balloon expands
the air particles are compressed; i.e., air pressure is momentarily higher than atmospheric. Conversely, when the
balloon contracts the air particles are sucked inward, resulting in rarefaction. The alternating compression and
rarefaction waves are propagated outward in all directions form the source. Only two of the three dimensions are
shown here; that is, the shape of the pressure disturbance is actually spherical rather than the circular pattern that is
shown here. Superimposed on the figure, in the graph labeled “one line of propagation,” is the resulting air pressure
waveform. Note that the pressure waveform takes on a high value during instants of compression and a low value
during instants of rarefaction. The figure also gives some idea of where the term uniform circular motion comes
from. If one were to make a graph plotting the height of the connecting rod on the rotating wheel as a function of
time it would trace out a perfect sinusoid; i.e., with exactly the shape of the pressure waveform that is superimposed
on the figure.

The Sound Pressure Waveform

Returning to Figure 3-7 for a moment, imagine that we chose some specific distance from the tuning fork to
observe how the movement and density of air particles varied with time. We would see individual air particles
oscillating small distances back and forth, and if we monitored particle density we would find that high particle
density (high air pressure) would be followed a moment later by relatively even particle spacing (atmospheric
pressure), which would be followed by a moment later by wide particle spacing (low air pressure), and so on.

Therefore, for an object that is vibrating sinusoidally, a graph showing variations in instantaneous air pressure

over time would also be sinusoidal. This is illustrated in Figure 3-9.
The vibratory patterns that have been discussed so far have all been sinusoidal. The concept of a sinusoid has
not been formally defined, but for our purposes it is enough to know that a sinusoid has precisely the smooth shape
that is shown in Figures such as 3-4 and 3-5. While sinusoids, also known as pure tones, have a very special place
in acoustic theory, they are rarely encountered in nature. The sound produced by a tuning fork comes quite close to a
sinusoidal shape, as do the simple tones that are used in hearing tests. Much more common in both speech and music
are more complex, nonsinusoidal patterns, to be discussed below. As will be seen in later chapters, these complex
vibratory patterns play a very important role in speech.

The Frequency Domain
We now arrive at what is probably the single most important concept for understanding both hearing and speech
acoustics. The graphs that we have used up to this point for representing either vibratory motion or the air pressure
disturbance created by this motion are called time domain representations. These graphs show how instantaneous
displacement (or instantaneous air pressure) varies over time. Another method for representing either sound or
vibration is called a frequency domain representation, also known as a spectrum. There are, in fact, two kinds of
frequency domain representations that are used to characterize sound. One is called an amplitude spectrum (also
known as a magnitude spectrum or a power spectrum, depending on how the level of the signal is represented)
and the other is called a phase spectrum. For reasons that will become clear soon, the amplitude spectrum is by far
the more important of the two. An amplitude spectrum is simply a graph showing what frequencies are present with
what amplitudes. Frequency is given along the x axis and some measure of amplitude is given on the y axis. A phase
spectrum is a graph showing what frequencies are present with what phases.
Figure 3-10 shows examples of the amplitude and phase spectra for several sinusoidal signals. The top panel

shows a time-domain representation of a sinusoid with a period of 10 ms and, consequently, a frequency of 100 Hz
(f = 1/t = 1/0.01 sec = 100 Hz). The peak-to-peak amplitude for this signal is 400 μPa, and the signal has a phase of
90 o. Since the amplitude spectrum is a graph showing what frequencies are present with what amplitudes, the
amplitude spectrum for this signal will show a single line at 100 Hz with a height of 400 μPa. The phase spectrum is
a graph showing what frequencies are present with what phases, so the phase spectrum for this signal will show a
single line at 100 Hz with a height of 90 o
. The second panel in Figure 3-10 shows a 200 Hz sinusoid with a peak-to-
peak amplitude of 200 μPa and a phase of 180 o
. Consequently, the amplitude spectrum will show a single line at 200
Hz with a height of 100 μPa, while the phase spectrum will show a line at 200 Hz with a height of 180 o.

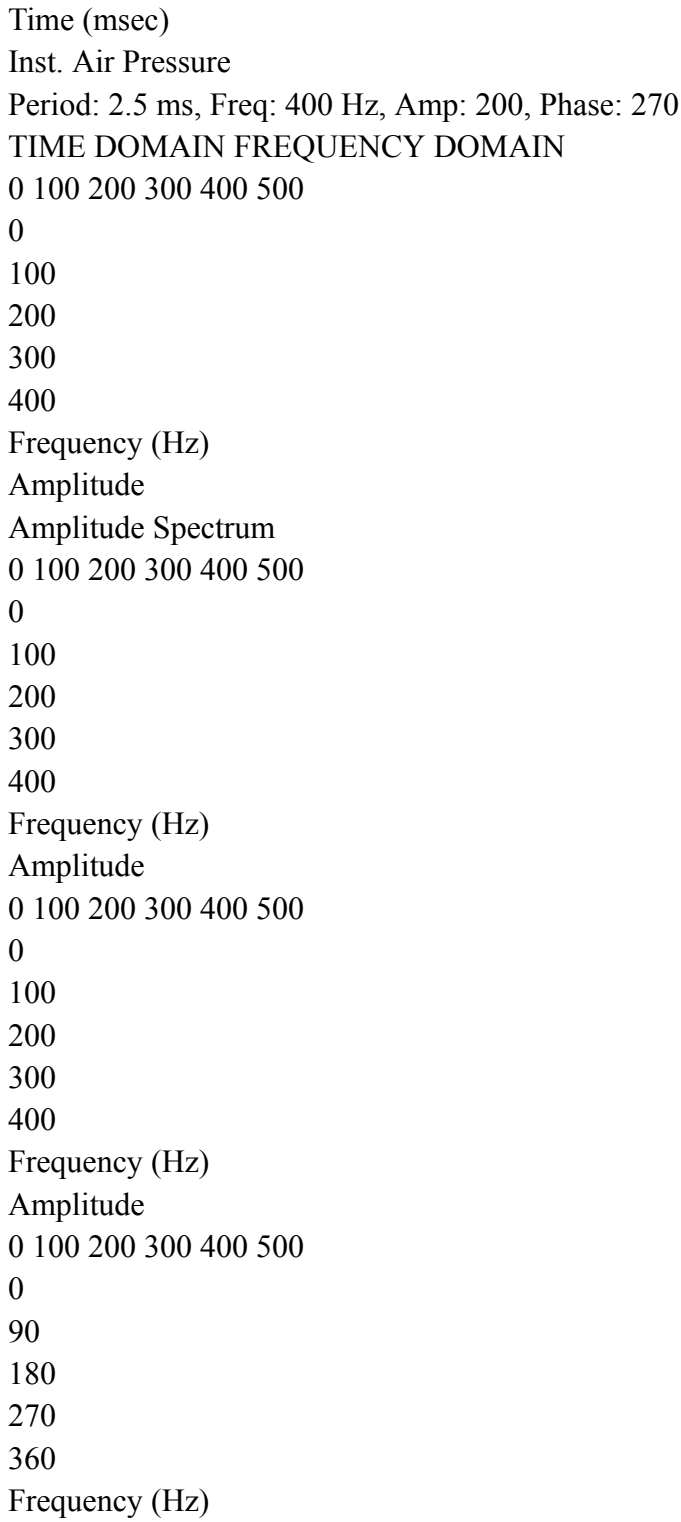
Complex Periodic Sounds

Sinusoids are sometimes referred to as simple periodic signals. The term "periodic" means that there is a
pattern that repeats itself, and the term "simple" means that there is only one frequency component present. This is
confirmed in the frequency domain representations in Figure 3-10, which all show a single frequency component in
both the amplitude and phase spectra. Complex periodic signals involve the repetition of a nonsinusoidal pattern,
and in all cases, complex periodic signals consist of more than a single frequency component. All nonsinusoidal
periodic signals are considered complex periodic.

Figure 3-8 Illustration of the propagation of a sound wave in two dimensions.

Figure 3-11 shows several examples of complex periodic signals, along with the amplitude spectra for these signals.
The time required to complete one cycle of the complex pattern is called the fundamental period. This is precisely
the same concept as the term period that was introduced earlier. The only reason for using the term "fundamental
period" instead of the simpler term "period" for complex periodic signals is to differentiate the fundamental period
(the time required to complete one cycle of the pattern as a whole) from other periods that may be present in the

signal (e.g., more rapid oscillations that might be observed within each cycle). The symbol for fundamental period is
t o. Fundamental frequency (f o) is calculated from fundamental period using the same kind of formula that we used
earlier for sinusoids:
fo = 1/to
The signal in the top panel of Figure 3-11 has a fundamental period of 5 ms, so fo = 1/0.005 = 200 Hz.
Examination of the amplitude spectra of the signals in Figure 3-11 confirms that they do, in fact, consist of
more than a single frequency. In fact, complex periodic signals show a very particular kind of amplitude spectrum
called a harmonic spectrum. A harmonic spectrum shows energy at the fundamental frequency and at whole
number multiples of the fundamental frequency. For example, the signal in the top panel of Figure 3-11 has energy
present at 200 Hz, 400 Hz, 600 Hz, 800 Hz, 1,000 Hz, 1200 Hz, and so on. Each frequency component in the
0 5 10 15 20 25 30
-200
-100
0
100
200
Inst. Air Pressure
Period: 10 ms, Freq: 100 Hz, Amp: 400, Phase: 90
0 5 10 15 20 25 30
-200
-100
0
100
200
Inst. Air Pressure
Period: 5 ms, Freq: 200 Hz, Amp: 200, Phase: 180
0 5 10 15 20 25 30
-200
-100
0
100
200

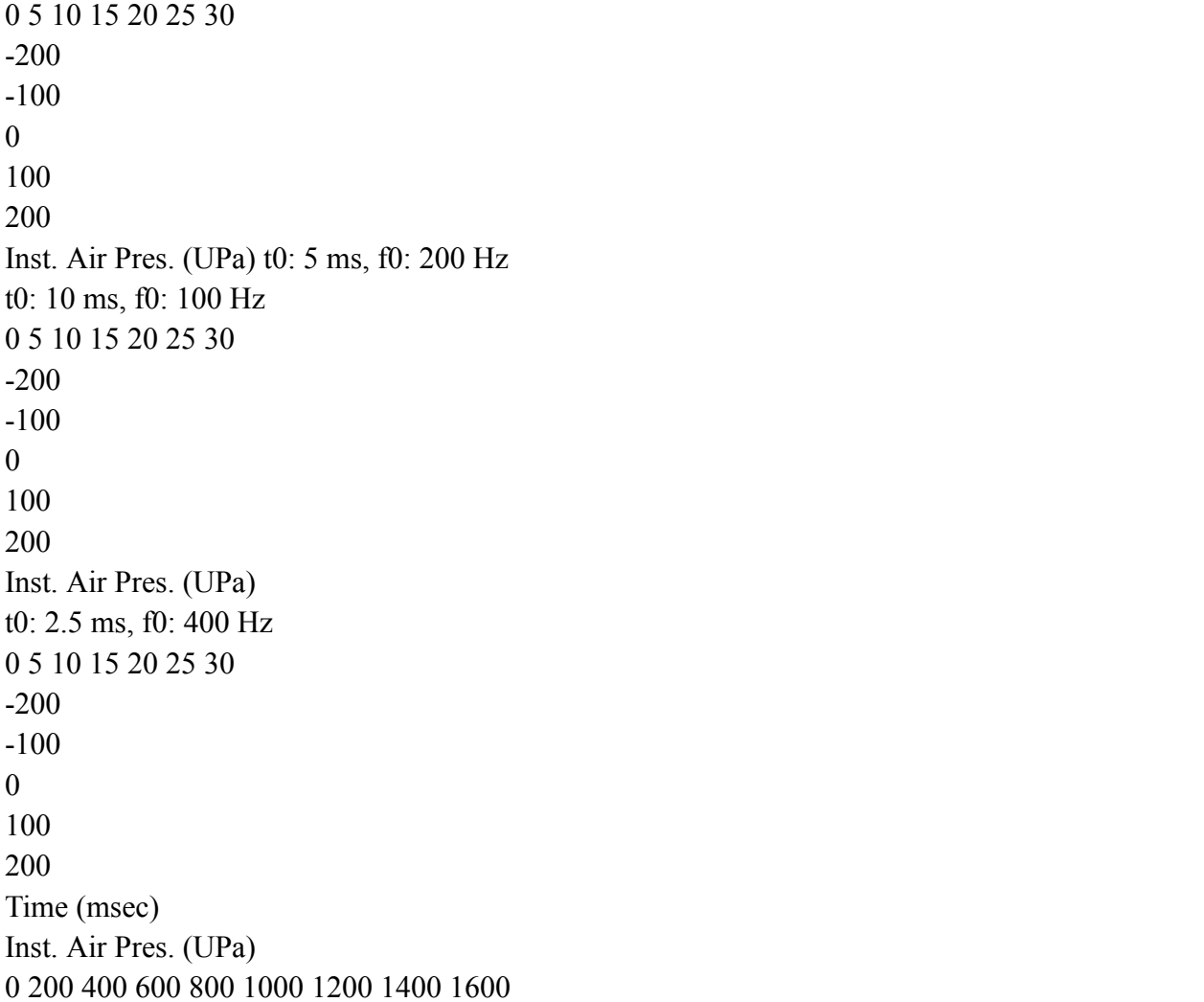

Time (msec)
Inst. Air Pressure
Period: 2.5 ms, Freq: 400 Hz, Amp: 200, Phase: 270
TIME DOMAIN FREQUENCY DOMAIN
0 100 200 300 400 500
0
100
200
300
400
Frequency (Hz)
Amplitude
Amplitude Spectrum
0 100 200 300 400 500
0
100
200
300
400
Frequency (Hz)
Amplitude
0 100 200 300 400 500
0
100
200
300
400
Frequency (Hz)
Amplitude
0 100 200 300 400 500
0
90
180
270
360
Frequency (Hz)
Phase
Phase Spectrum
0 100 200 300 400 500
0

90
180
270
360
Frequency (Hz)
Phase
0 100 200 300 400 500
0
90
180
270
360
Frequency (Hz)
Phase
Figure 3-10. Time and frequency domain representations of three sinusoids. The frequency domain
consists of two graphs: an amplitude spectrum and a phase spectrum. An amplitude spectrum is a graph showing what frequencies are present with what amplitudes, and a phase spectrum is a graph
showing the phases of each frequency component.

amplitude spectrum of a complex periodic signal is called a harmonic (also known as a partial). The fundamental
frequency, in this case 200 Hz, is also called the first harmonic, the 400 Hz component (2 · fo) is called the second
harmonic, the 600 Hz component (3 · fo) is called the third harmonic, and so on.
The second panel in Figure 3-11 shows a complex periodic signal with a fundamental period of 10 ms and,
consequently, a fundamental frequency of 100 Hz. The harmonic spectrum that is associated with this signal will
therefore show energy at 100 Hz, 200 Hz, 300 Hz, 400 Hz, 500 Hz, and so on. The bottom panel of Figure 3-11
shows a complex periodic signal with a fundamental period of 2.5 ms, a fundamental frequency of 400 Hz, and
harmonics at 400, 800, 1200, 1600, and so on. Notice that there two completely interchangeable ways to define the
term fundamental frequency. In the time domain, the fundamental frequency is the number of cycles of the complex
pattern that are completed in one second. In the frequency domain, except in the case of certain special signals, the

fundamental frequency is the lowest harmonic in the harmonic spectrum. Also, the fundamental frequency defines
the harmonic spacing; that is, when the fundamental frequency is 100 Hz, harmonics will be spaced at 100 Hz

Figure 3-11. Time and frequency domain representations of three complex periodic signals. Complex periodic signals have harmonic spectra, with energy at the fundamental frequency (f0) and
at whole number multiples of f0 (f0. 2, f0. 3, f0. 4, etc.) For example, the signal in the upper left, with a
fundamental frequency of 200 Hz, shows energy at 200 Hz, 400 Hz, 600 Hz, etc. In the spectra on
the right, amplitude is measured in arbitrary units. The main point being made in this figure is the
distribution of harmonic frequencies at whole number multiples of f0 for complex periodic signals.



0
20
40
60
80
100
120
Frequency (Hz)
Amplitude
0 200 400 600 800 1000 1200 1400 1600
0
20
40
60
80
100
120
Frequency (Hz)
Amplitude
0 200 400 600 800 1000 1200 1400 1600
0
20
40
60
80
100
120
Frequency (Hz)
Amplitude
TIME DOMAIN FREQUENCY DOMAIN
The Physics of Sound 11
0 10 20 30 40 50
-200
-100
0
100
200
Inst. Air Pres. (UPa)
White Noise
/s/

0 10 20 30 40 50
-200
-100
0
100
200
Inst. Air Pres. (UPa)
/f/
0 10 20 30 40 50
-200
-100
0
100
200
TIME (msec)
Inst. Air Pres. (UPa)
0 1 2 3 4 5 6 7 8 9 10
0
20
40
60
80
100
Amplitude
0 1 2 3 4 5 6 7 8 9 10
0
20
40
60
80
100
Amplitude
0 1 2 3 4 5 6 7 8 9 10
0
20
40
60
80
100
Frequency (kHz)

Amplitude
TIME DOMAIN FREQUENCY DOMAIN
intervals (i.e., 100, 200, 300 ...), when the fundamental frequency is 125 Hz, harmonics will be spaced at 125 Hz
intervals (i.e., 125, 250, 375...), and when the fundamental frequency is 200 Hz, harmonics will be spaced at 200 Hz
intervals (i.e., 200, 400, 600 ...). (For some special signals this will not be the case.2) So, when fo is low, harmonics
will be closely spaced, and when fo is high, harmonics will be widely spaced. This is clearly seen in Figure 3-11: the
signal with the lowest f0 (100 Hz, the middle signal) shows the narrowest harmonic spacing, while the signal with
the highest f0 (400 Hz, the bottom signal) shows the widest harmonic spacing.
There are certain characteristics of the spectra of complex periodic sounds that can be determined by making simple
measurements of the time domain signal, and there are certain other characteristics that require a more complex
analysis. For example, simply by examining the signal in the bottom panel of Figure 3-11 we can determine that it is
complex periodic (i.e., it is periodic but not sinusoidal) and therefore it will show a harmonic spectrum with energy
at whole number multiples of the fundamental frequency. Further, by measuring the fundamental period (2.5 ms)
2There are some complex periodic signals that have energy at odd multiples of the fundamental frequency only. A square wave, for
example, is a signal that alternates between maximum positive amplitude and maximum negative amplitude. The spectrum of square wave shows
energy at odd multiples of the fundamental frequency only. Also, a variety of simple signal processing tricks can be used to create signals with
harmonics at any arbitrary set of frequencies. For example, it is a simple matter to create a signal with energy at 400, 500, and 600 Hz only.
While these kinds of signals can be quite useful for conducting auditory perception experiments, it remains true that most naturally occurring
complex periodic signals have energy at all whole number multiples of the fundamental frequency.
Figure 3-12. Time and frequency domain representations of three non-transient complex aperiodic
signals. Unlike complex periodic signals, complex aperiodic signals show energy that is spread across the spectrum. This type of spectrum is called dense or continuous. These spectra have a very

different appearance from the “picket fence” look that is associated with the discrete, harmonic spectra of complex periodic signals.

and converting it into fundamental frequency (400 Hz), we are able to determine that the signal will have energy at
400, 800, 1200, 1600, etc. But how do we know the amplitude of each of these frequency components? And how do
we know the phase of each component? The answer is that you cannot determine harmonic amplitudes or phases
simply by inspecting the signal or by making simple measurements of the time domain signals with a ruler. We will
see soon that a technique called Fourier analysis is able to determine both the amplitude spectrum and the phase
spectrum of any signal. We will also see that the inner ears of humans and many other animals have developed a
trick that is able to produce a neural representation that is comparable in some respects to an amplitude spectrum.
We will also see that the ear has no comparable trick for deriving a representation that is equivalent to a phase
spectrum. This explains why the amplitude spectrum is far more important for speech and hearing applications than
the phase spectrum. We will return to this point later.
To summarize: (1) a complex periodic signal is any periodic signal that is not sinusoidal, (2) complex periodic
signals have energy at the fundamental frequency (fo) and at whole number multiples of the fundamental frequency
($2 \cdot fo$, $3 \cdot fo$, $4 \cdot fo$ ...), and (3) although measuring the fundamental frequency allows us to determine the frequency
locations of harmonics, there is no simple measurement that can tell us harmonic amplitudes or phases. For this,
Fourier analysis or some other spectrum analysis technique is needed.
Figure 3-13. Time and frequency domain representations of three transients. Transients are complex
aperiodic signals that are defined by their brief duration. Pops, clicks, and the sound gun fire are examples of transients. In common with longer duration complex aperiodic signals, transients show
dense or continuous spectra, very unlike the discrete, harmonic spectra associated with complex periodic
d
0 10 20 30 40 50 60 70 80 90 100

-200
-100
0
100
200
Inst. Amp. (UPa)
Rap on Desk
Clap
0 10 20 30 40 50 60 70 80 90 100
-200
-100
0
100
200
Inst. Amp. (UPa)
Tap on Cheek
0 10 20 30 40 50 60 70 80 90 100
-200
-100
0
100
200
TIME (msec)
Inst. Amp. (UPa)
0 1 2 3 4 5
0
20
40
60
80
100
Amplitude
0 1 2 3 4 5
0
20
40
60
80
100
Amplitude

0 1 2 3 4 5
0
20
40
60
80
100
Frequency (kHz)
Amplitude
TIME DOMAIN FREQUENCY DOMAIN
The Physics of Sound 13
-200
0
200
Inst. Air Pres.
(a)
-200
0
200
Inst. Air Pres.
(b)
-200
0
200
Inst. Air Pres.
(c)
-200
0
200
Inst. Air Pres.
(d)
-300
0
300
Inst. Air Pres.
(e)
Time ->
Aperiodic Sounds
An aperiodic sound is any sound that does not show a repeating pattern in its time domain representation. There are

many aperiodic sounds in speech. Examples include the hissy sounds associated with fricatives such as /f/ and /s/,
and the various hisses and pops associated with articulatory release for the stop consonants /b,d,g,p,t,k/. Examples of
non-speech aperiodic sounds include a drummer's cymbal or snare drum, the hiss produced by a radiator, and static
sound produced by a poorly tuned radio. There are two types of aperiodic sounds: (1) continuous aperiodic sounds
(also known as noise) and (2) transients. Although there is no sharp cutoff, the distinction between continuous
aperiodic sounds and transients is based on duration. Transients (also "pops" and "clicks") are defined by their very
brief duration, and continuous aperiodic sounds are of longer duration. Figure 3-12 shows several examples of time
domain representations and amplitude spectra for continuous aperiodic sounds. The lack of periodicity in the time
Figure 3-14. Illustration of the principle underlying Fourier analysis. The complex periodic signal
shown in panel e was derived by point-for-point summation of the sinusoidal signals shown in panels a-d. Point-for-point summation simply means beginning at time zero (i.e., the start of the signal) and adding the instantaneous amplitude of signal a to the instantaneous amplitude of signal b
at time zero, then adding that sum to the instantaneous amplitude of signal c, also at time zero, then
adding that sum to instantaneous amplitude of signal d at time zero. The sum of instantaneous amplitudes at time zero of signals a-d is the instantaneous amplitude of the composite signal e at time zero. For example, at time zero the amplitudes of sinusoids a-d are 0, +100, -200, and 0, respectively, producing a sum of -100. This agrees with the instantaneous amplitude at the very beginning of composite signal e. The same summation procedure is followed for all time points.

domain is quite evident; that is, unlike the periodic sounds we have seen, there is no pattern that repeats itself over
time.
All aperiodic sounds -- both continuous and transient -- are complex in the sense that they always consist of
energy at more than one frequency. The characteristic feature of aperiodic sounds in the frequency domain is a
dense or continuous spectrum, which stands in contrast to the harmonic spectrum that is associated with complex

periodic sounds. In a harmonic spectrum, there is energy at the fundamental frequency, followed by a gap with little
or no energy, followed by energy at the second harmonic, followed by another gap, and so on. The spectra of
aperiodic sounds do not share this "picket fence" appearance. Instead, energy is smeared more-or-less continuously
across the spectrum. The top panel in Figure 3-12 shows a specific type of continuous aperiodic sound called white
noise. By analogy to white light, white noise has a flat amplitude spectrum; that is, approximately equal amplitude at
all frequencies. The middle panel in Figure 3-12 shows the sound /s/, and the bottom panel shows sound /f/. Notice
that the spectra for all three sounds are dense; that is, they do not show the "picket fence" look that reveals harmonic
structure. As was the case for complex periodic sounds, there is no way to tell how much energy there will be at
different frequencies by inspecting the time domain signal or by making any simple measures with a ruler. Likewise,
there is no simple way to determine the phase spectrum. So, after inspecting a time-domain signal and determining
that it is aperiodic, all we know for sure is that it will have a dense spectrum rather than a harmonic spectrum.
Figure 3-13 shows time domain representations and amplitude spectra for three transients. The transient in the
top panel was produced by rapping on a wooden desk, the second is a single clap of the hands, and the third was
produced by holding the mouth in position for the vowel /o/, and tapping the cheek with an index finger. Note the
brief durations of the signals. Also, as with continuous aperiodic sounds, the spectra associated with transients are
dense; that is, there is no evidence of harmonic organization. In speech, transients occur at the instant of articulatory
release for stop consonants. There are also some languages, such as the South African languages Zulu, Hottentot,
and Xhosa, that contain mouth clicks as part of their phonemic inventory (MacKay, 1986).
Fourier Analysis
TIME DOMAIN
Time ->
Inst. Air Pres.
Fourier

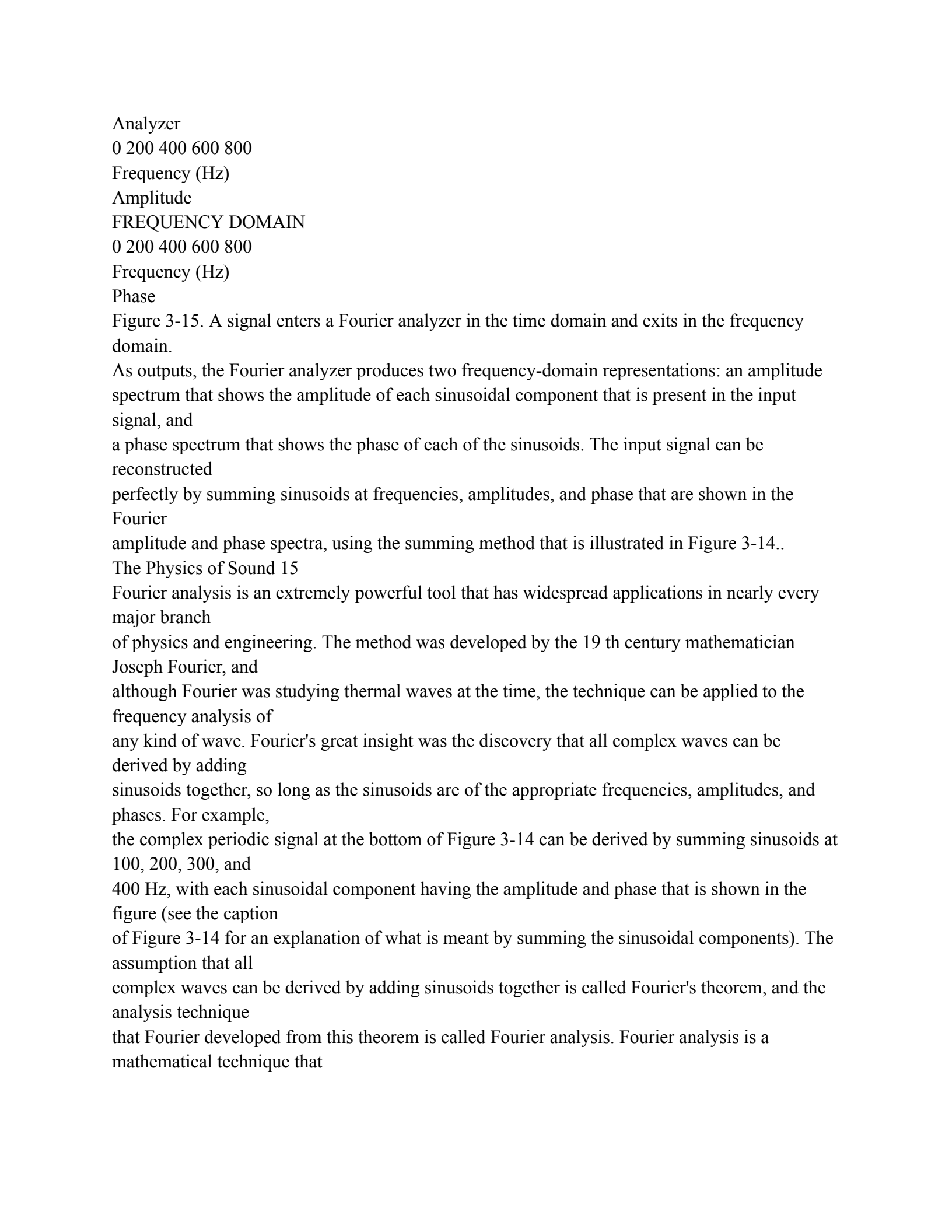


Figure 3-15. A signal enters a Fourier analyzer in the time domain and exits in the frequency domain.

As outputs, the Fourier analyzer produces two frequency-domain representations: an amplitude spectrum that shows the amplitude of each sinusoidal component that is present in the input signal, and a phase spectrum that shows the phase of each of the sinusoids. The input signal can be reconstructed perfectly by summing sinusoids at frequencies, amplitudes, and phase that are shown in the Fourier amplitude and phase spectra, using the summing method that is illustrated in Figure 3-14..

Fourier analysis is an extremely powerful tool that has widespread applications in nearly every major branch of physics and engineering. The method was developed by the 19 th century mathematician Joseph Fourier, and although Fourier was studying thermal waves at the time, the technique can be applied to the frequency analysis of any kind of wave. Fourier's great insight was the discovery that all complex waves can be derived by adding sinusoids together, so long as the sinusoids are of the appropriate frequencies, amplitudes, and phases. For example, the complex periodic signal at the bottom of Figure 3-14 can be derived by summing sinusoids at 100, 200, 300, and 400 Hz, with each sinusoidal component having the amplitude and phase that is shown in the figure (see the caption of Figure 3-14 for an explanation of what is meant by summing the sinusoidal components). The assumption that all complex waves can be derived by adding sinusoids together is called Fourier's theorem, and the analysis technique that Fourier developed from this theorem is called Fourier analysis. Fourier analysis is a mathematical technique that

takes a time domain signal as its input and determines: (1) the amplitude of each sinusoidal component that is
present in the input signal, and (2) the phase of each sinusoidal component that is present in the input signal.
Another way of stating this is that Fourier analysis takes a time domain signal as its input and produces two
frequency domain representations as output: (1) an amplitude spectrum, and (2) a phase spectrum.
The basic concept is illustrated in Figure 3-15, which shows a time domain signal entering the Fourier analyzer.
Emerging at the output of the Fourier analyzer is an amplitude spectrum (a graph showing the amplitude of each
sinusoid that is present in the input signal) and a phase spectrum (a graph showing the phase of each sinusoid that is
present in the input signal). The amplitude spectrum tells us that the input signal contains: (1) 200 Hz sinusoid with
an amplitude of 100 μPa, a 400 Hz sinusoid with an amplitude of 200 μPa, and a 600 Hz sinusoid with an amplitude
of 50 μPa. Similarly, the phase spectrum tells us that the 200 Hz sinusoid has a phase of 90 o, the 400 Hz sinusoid
has a phase of 180 o, and the 600 Hz sinusoid has a phase of 270 o. If Fourier's theorem is correct, we should be able
to reconstruct the input signal by summing sinusoids at 200, 400, and 600 Hz, using the amplitudes and phases that
are shown. In fact, summing these three sinusoids in this way would precisely reproduce the original time domain
signal; that is, we would get back an exact replica of our original signal, and not just a rough approximation to it.
For our purposes it is not important to understand how Fourier analysis works. The most important point about
Fourier's idea is that, visual appearances aside, all complex waves consist of sinusoids of varying frequencies,
amplitudes, and phases. In fact, Fourier analysis applies not only to periodic signals such as those shown in Figure
3-15, but also to noise and transients. In fact, the amplitude spectra of the aperiodic signals shown in Figure 3-13
were calculated using Fourier analysis. In later chapters we will see that the auditory system is able to derive a
neural representation that is roughly comparable to a Fourier amplitude spectrum. However, as was mentioned

earlier, the auditory system does not derive a representation comparable to a Fourier phase spectrum. As a result,
listeners are very sensitive to changes in the amplitude spectrum but are relatively insensitive to changes in phase.

Some Additional Terminology

Overtones vs. Harmonics: The term overtone and the term harmonic refer to the same concept; they are just
counted differently. As we have seen, in a harmonic series such as 100, 200, 300, 400, etc., the 100 Hz component
can be referred to as either the fundamental frequency or the first harmonic; the 200 Hz component is the second
harmonic, the 300 Hz component is the third harmonic, and so on. An alternative set of terminology would refer to
the 100 Hz component as the fundamental frequency, the 200 Hz component as the first overtone, the 300 Hz
component as the second overtone, and so on. Use of the term overtone tends to be favored by those interested in
musical acoustics, while most other acousticians tend to use the term harmonic.

Octaves vs. Harmonics: An octave refers to a doubling of frequency. So, if we begin at 100 Hz, the next octave up
would 200 Hz, the next would be 400 Hz, the next would be 800 Hz, and so on. Note that this is quite different from
a harmonic progression. A harmonic progression beginning at 300 Hz would be 300, 600, 900, 1200, 1500, etc.,
while an octave progression would be 300, 600, 1200, 2400, 4800, etc. There is something auditorilly natural about
octave spacing, and octaves play a very important role in the organization of musical scales. For example, on a piano
keyboard, middle A (A5) is 440 Hz, A above middle A (A 6) is 880 Hz, A7 is 1,760 and so on. (See Box 3-2).

Wavelength: The concept of wavelength is best illustrated with an example given by Small (1973). Small asks us
to imagine dipping a finger repeatedly into a puddle of water at a perfectly regular interval. Each time the finger hits
the water, a wave is propagated outward, and we would see a pattern formed consisting of a series of concentric

circles (see Figure 3-16). Wavelength is simply the distance between the adjacent waves. Precisely the same concept

can be applied to sound waves: wavelength is simply the distance between one compression wave and the next (or
one rarefaction wave and the next or, more generally, the distance between any two corresponding points in adjacent
waves). For our purposes, the most important point to be made about wavelength is that there is a simple
relationship between frequency and wavelength. Using the puddle example, imagine that we begin by dipping our
finger into the puddle at a very slow rate; that is, with a low "dipping frequency." Since the waves have a long
period of time to travel from one dip to the next, the wavelength will be large. By the same reasoning, the
wavelength becomes smaller as the "dipping frequency" is increased; that is, the time allowed for the wave to travel
at high "dipping frequency" is small, so the wavelength is small. Wavelength is a measure of distance, and the
formula for calculating wavelength is a straightforward algebraic rearrangement of the familiar "distance = rate ·
time" formula from junior high school.
$\lambda = c/f$, where: λ = wavelength
c = the speed of sound
f = frequency
By rearranging the formula, frequency can be calculated if wavelength and the speed of sound are known:
$f = c/\lambda$
Lower Frequency
(Longer Wavelength)
Higher Frequency
(Shorter Wavelength)
Figure 3-16. Wavelength is a measure of the distance between the crest of one cycle of a wave and the
crest of the next cycle (or trough to trough or, in fact, the distance between any two corresponding
points in the wave). Wavelength and frequency are related to one another. Because the wave has only a
short time to travel from one cycle to the next, high frequencies produce short wavelengths.
Conversely, because of the longer travel times, low frequencies produce long wavelengths.

Spectrum Envelope: The term spectrum envelope refers to an imaginary smooth line drawn to enclose an

amplitude spectrum. Figure 3-17 shows several examples. This is a rather simple concept that will play a very important role in understanding certain aspects of auditory perception. For example, we will see that our perception of a perceptual attribute called timbre (also called sound quality) is controlled primarily by the shape of the spectrum envelope, and not by the fine details of the amplitude spectrum. The examples in Figure 3-17 show how differences in spectrum envelope play a role in signaling differences in one specific example of timbre called vowel quality (i.e., whether a vowel sounds like /i/ vs. /a/ vs. /u/, etc.). For example, panels a and b in Figure 3-17 show the vowel /å/ produced at two different fundamental frequencies. (We know that the fundamental frequencies are different because one spectrum shows wide harmonic spacing and the other shows narrow harmonic spacing.)

The fact that the two vowels are heard as /a/ despite the difference in fundamental frequency can be attributed to the fact that these two signals have similar spectrum envelopes. Panels c and d in Figure 3-17 show the spectra of two signals with different spectrum envelopes but the same fundamental frequency (i.e., with the same harmonic spacing). As we will see in the chapter on auditory perception, differences in fundamental frequency are perceived as differences in pitch. So, for signals (a) and (b) in Figure 3-17, the listener will hear the same vowel produced at two different pitches. Conversely, for signals (c) and (d) in Figure 3-17, the listener will hear two different vowels produced at the same pitch. We will return to the concept of spectrum envelope in the chapter on auditory perception.

Amplitude Envelope: The term amplitude envelope refers to an imaginary smooth line that is drawn on top of a time domain signal. Figure 3-18 shows sinusoids that are identical except for their amplitude envelopes. It can be seen that the different amplitude envelopes reflect differences in the way the sounds are turned on and off. For example, panel a shows a signal that is turned on abruptly and turned off abruptly; panel b shows a signal that is

turned on gradually and turned off abruptly; and so on. Differences in amplitude envelope have an important effect
on the quality of a sound. As we will see in the chapter on auditory perception, amplitude envelope, along with
spectrum envelope discussed above, is another physical parameter that affects timbre or sound quality. For
0 1 2 3
0
10
20
30
40
50
60
70
Frequency (kHz)
Amplitude
(a) Vowel: /a/, f0: 100 Hz
0 1 2 3
0
10
20
30
40
50
60
70
Frequency (kHz)
Amplitude
(b) Vowel: /a/, f0: 200 Hz
0 1 2 3
0
10
20
30
40
50
60
70
Frequency (kHz)

Amplitude
(c)
Vowel: /i/, f0: 150 Hz
0 1 2 3
0
10
20
30
40
50
60
70
Frequency (kHz)
Amplitude
(d)
Vowel: /u/, f0: 150 Hz
Figure 3-17. A spectrum envelope is an imaginary smooth line drawn to enclose an amplitude spectrum. Panels a and b show the spectra of two signals (the vowel /ɑ/) with different fundamental
frequencies (note the differences in harmonic spacing) but very similar spectrum envelopes. Panels c
and d show the spectra of two signals with different spectrum envelopes (the vowels /i/ and /u/ in this
case) but the same fundamental frequencies (i.e., the same harmonic spacing).

example, piano players know that a given note will sound different depending on whether or not the damping pedal
is used. Similarly, notes played on a stringed instrument such as a violin or cello will sound different depending on
whether the note is plucked or bowed. In both cases, the underlying acoustic difference is amplitude envelope.
Acoustic Filters
As will be seen in subsequent chapters, acoustic filtering plays a central role in the processing of sound by the
inner ear. The human vocal tract also serves as an acoustic filter that modifies and shapes the sounds that are created
by the larynx and other articulators. For this reason, it is quite important to understand how acoustic filters work. In
the most general sense, the term filter refers to a device or system that is selective about the kinds of things that are

allowed to pass through versus the kinds of things that are blocked. An oil filter, for example, is designed to allow
oil to pass through while blocking particles of dirt. Of special interest to speech and hearing science are frequency
selective filters. These are devices that allow some frequencies to pass through while blocking or attenuating other
frequencies. (The term attenuate means to weaken or reduce in amplitude).
A simple example of a frequency selective filter from the world of optics is a pair of tinted sunglasses. A piece
of white paper that is viewed through red tinted sunglasses will appear red. Since the original piece of paper is
white, and since we know that white light consists of all of the visible optical frequencies mixed in equal amounts,
the reason that the paper appears red through the red tinted glasses is that optical frequencies other than those
corresponding to red are being blocked or attenuated by the optical filter. As a result, it is primarily the red light that
is being allowed to pass through. (Starting at the lowest optical frequency and going to the highest, light will appear
red, orange, yellow, green, blue, indigo, and violet.)
Inst. Air Pres.
(a)
Signals Differing in Amplitude Envelope
Inst. Air Pres.
(b)
Inst. Air Pres.
(c)
Time ->
Inst. Air Pres.
(d)
Figure 3-18. Amplitude envelope is an imaginary smooth line drawn to enclose a time-domain signal.
This feature describes how a sound is turned on and turned off; for example, whether the sound is
turned on abruptly and turned off abruptly (panel a), turned on gradually and turned off abruptly (panel
b), turned on abruptly and turned off gradually (panel c), or turned on and off gradually (panel d).

A graph called a frequency response curve is used to describe how a frequency selective filter will behave. A
frequency response curve is a graph showing how energy at different frequencies will be affected by the filter.
Specifically, a frequency response curve plots a variable called "gain" as a function of variations in the frequency of
the input signal. Gain is the amount of amplification provided by the filter at different signal frequencies. Gains are
interpreted as amplitude multipliers; for example, suppose that the gain of a filter at 100 Hz is 1.3. If a 100 Hz
sinusoid enters the filter measuring 10 uPa, the amplitude at the output of the filter at 100 Hz will measure 13 μPa
(10 μPa x 1.3 = 13 μPa). The only catch in this scheme is that gains can and very frequently are less than 1, meaning
that the effect of the filter will be to attenuate the signal. For example, if the gain at 100 Hz is 0.5, a 10 μPa input
signal at 100 Hz will measure 5 μPa at the output of the filter. When the filter gain is 1.0, the signal is unaffected by
the filter; i.e., a 10 μPa input signal will measure 10 μPa at the output of the filter.
Figure 3-19 shows frequency response curves for several optical filters. Panel a shows a frequency response
curve for the red optical filter discussed in the example above. If we put white light into the filter in panel a, the
signal amplitude at the output of the filter will be high only when the frequency of the input signal is low. This is
because the gain of the filter is high only in the low-frequency portion of the frequency-response curve. This is an
example of a lowpass filter; that is, a filter that allows low frequencies to pass through. Panel b shows an optical
filter that has precisely the reverse effect on an input signal; that is, this filter will allow high frequencies to pass
through while attenuating low- and mid-frequency signals. A white surface viewed through this filter would
therefore appear violet. This is an example of a highpass filter. Panel c shows the frequency response curve for a
filter that allows a band of energy in the center of the spectrum to pass through while attenuating signal components
of higher and lower frequency. A white surface viewed through this filter would appear green. This is called a
bandpass filter.

Acoustic filters do for sound exactly what optical filters do for light; that is, they allow some frequencies to pass
through while attenuating other frequencies. To get a better idea of how a frequency response curve is measured,
imagine that we ask a singer to attempt to shatter a crystal wine glass with a voice signal alone. To see how the
frequency response curve is created we have to make two rather unrealistic assumptions: (1) we need to assume that
the singer is able to produce a series of pure tones of various frequencies (the larynx, in fact, produces a complex
periodic sound and not a sinusoid), and (2) the amplitudes of these pure tones are always exactly the same. The wine
glass will serve as the filter whose frequency response curve we wish to measure. As shown in Figure 3-20, we
attach a vibration meter to the wine glass, and the reading on this meter will serve as our measure of output
Figure 3-19. Frequency response curves for three optical filters. The lowpass filter on the left allows
low frequencies to pass through, while attenuating or blocking optical energy at higher frequencies.
The highpass filter in the middle has the opposite effect, allowing high frequencies to pass through,
while attenuating or blocking optical energy at lower frequencies. The bandpass filter on the right
allows a band of optical frequencies in the center of the spectrum to pass through, while attenuating or
blocking energy at higher and lower frequencies.

amplitude for the filter. For the purpose of this example, will assume that the signal frequency needed to break the
glass is 500 Hz. We now ask the singer to produce a low frequency signal, say 50 Hz. Since this frequency is quite
remote from the 500 Hz needed to break the glass, the output amplitude measured by the vibration meter will be
quite low. As the singer gets closer and closer to the required 500 Hz, the measured output amplitude will increase
systematically until the glass finally breaks. If we assume that the glass does not break but rather reaches a
maximum amplitude just short of that required to shatter the glass, we can continue our measurement of the

frequency response curve by asking the singer to produce signals that are increasingly high in frequency. We would find that the output amplitude would become lower and lower the further we got from the 500 Hz natural vibrating frequency of the wine glass. The pattern that is traced by our measures of output amplitude at each signal frequency would resemble the frequency response curve we saw earlier for green sunglasses; that is, we would see the frequency response curve for a bandpass filter.

Additional Comments on Filters

Cutoff Frequency, Center Frequency, Bandwidth. The top panel of Figure 3-21 shows frequency response curves for two lowpass filters that differ in a parameter called cutoff frequency. Both filters allow low frequencies to pass through while attenuating high frequencies; the filters differ only in the frequency at which the attenuation begins.

The bottom panel of Figure 3-21 shows two highpass filters that differ in cutoff frequency. There are two additional terms that apply only to bandpass filters. In our wineglass example above, the natural vibrating frequency of the wine glass was 300 Hz. For this reason, when the frequency response curve is measured, we find that the wine glass reaches its maximum output amplitude at 300 Hz. This is called the center frequency or resonance of the filter. It is possible for two bandpass filters to have the same center frequency but differ with respect to a property called

Figure 3-20. Illustration of how the frequency response curve of a crystal wine glass might be measured. Our singer produces a series of sinusoids that are identical in amplitude but cover a wide range of frequencies. (This part of the example is unrealistic: the human larynx produces a complex sound rather than a sinusoid.) The gain of the wine glass filter can be traced out by measuring the amplitude of vibration at the different signal frequencies.)

bandwidth. Figure 3-22 shows two filters that differ in bandwidth. The tall, thin frequency response curve describes a narrow band filter. For this type of filter, output amplitude reaches a very sharp peak at the center frequency and drops off abruptly on either side of the peak. The other frequency response curve describes a wide band filter (also

called broad band). For the wide band filter, the peak that occurs at the resonance of the filter is less sharp and the
drop in output amplitude on either side of the center frequency is more gradual.
Fixed vs. Variable Filters. A fixed filter is a filter whose frequency response curve cannot be altered. For example,
an engineer might design a lowpass filter that attenuates at frequencies above 500 Hz, or a bandpass filter that passes
with a center frequency of 1,000 Hz. It is also possible to create a filter whose characteristics can be varied. For
example, the tuning dial on a radio controls the center frequency of a narrow bandpass filter that allows a single
radio channel to pass through while blocking channels at all other frequencies. The human vocal tract is an example
0 1000 2000 3000 4000
0.0
0.2
0.4
0.6
0.8
1.0
Frequency (Hz)
Gain
Lowpass Filters with Different
Cutoff Frequencies
0 1000 2000 3000 4000
0.0
0.2
0.4
0.6
0.8
1.0
Frequency (Hz)
Gain
Highpass Filters with Different
Cutoff Frequencies
Figure 3-21. Lowpass and highpass filters differing in cutoff frequency.
0 1000 2000 3000 4000
0.0
0.2
0.4

Figure 3-22. Frequency response curves for two bandpass filters with identical center frequencies but different bandwidths. Both filters pass a band of energy centered around 2000 Hz, but the narrow band filter is more selective than the wide band filter; that is, gain decreases at a higher rate above and below the center frequency for the narrow band filter than for the wide band filter

of a variable filter of the most spectacular sort. For example: (1) during the occlusion interval that occurs in the production of a sound like /b/, the vocal tract behaves like a lowpass filter; (2) in the articulatory posture for sounds like /s/ and /sh/ the vocal tract behaves like a highpass filter; and (3) in the production of vowels, the vocal tract behaves like a series of bandpass filters connected to one another, and the center frequencies of these filters can be adjusted by changing the positions of the tongue, lips, and jaw. To a very great extent, the production of speech involves making adjustments to the articulators that have the effect of setting the vocal tract filter in differ modes to produce the desired sound quality. We will have much more to say about this in later chapters.

Frequency Response Curves vs. Amplitude Spectra. It is not uncommon for students to confuse a frequency response curve with an amplitude spectrum. The axis labels are rather similar: an amplitude spectrum plots amplitude on the y axis and frequency on the x axis, while a frequency response curve plots gain on the y axis and frequency on the x axis. The apparent similarities are deceiving, however, since a frequency response curve and an amplitude spectrum display very different kinds of information. The difference is that an amplitude spectrum describes a sound while a frequency response curve describes a filter. For any given sound wave, an amplitude

spectrum tells us what frequencies are present with what amplitudes. A frequency response curve, on the other hand,
describes a filter, and for that filter, it tells us what frequencies will be allowed to pass through and what frequencies
will be attenuated. Keeping these two ideas separate will be quite important for understanding the key role played by
filters in both hearing and speech science.

Resonance

The concept of resonance has been alluded to on several occasions but has not been formally defined. The term
resonance is used in two different but very closely related ways. The term resonance refers to: (1) the phenomenon
of forced vibration, and (2) natural vibrating frequency (also resonant frequency or resonance frequency) To
gain an appreciation for both uses of this term, imagine the following experiment. We begin with two identical
tuning forks, each tuned to 435 Hz. Tuning fork A is set into vibration and placed one centimeter from tuning fork
B, but not touching it. If we now hold tuning fork B to a healthy ear, we will find that it is producing a 435 Hz tone
that is faint but quite audible, despite the fact that it was not struck and did not come into physical contact with
tuning fork A. The explanation for this "action-at-a-distance" phenomenon is that the sound wave generated by
tuning fork A forces tuning fork B into vibration; that is, the series of compression and rarefaction waves will
alternately push and pull the tuning fork, resulting in vibration at the frequency being generated by tuning fork A.

The phenomenon of forced vibration is not restricted to this "action-at-a-distance" case. The same effect can be
demonstrated by placing a vibrating tuning fork in contact with a desk or some other hard surface. The intensity of
the signal will increase dramatically because the tuning fork is forcing the desk to vibrate, resulting in a larger
volume of air being compressed and rarefied.3

Returning to our original tuning fork experiment, suppose that we repeat this test using two mismatched tuning
forks; for example, tuning fork A with a natural frequency of 256 Hz and tuning fork B with a natural vibrating

frequency of 435 Hz. If we repeat the experiment – setting tuning fork A into vibration and holding it one centimeter
from tuning fork B – we will find that tuning fork B does not produce an audible tone. The reason is that forced
vibration is most efficient when the frequency of the driving force is closest to the natural vibration frequency of the
object that is being forced to vibrate. Another way to think about this is that tuning fork B in these experiments is
behaving like a filter that is being driven by the signal produced by tuning fork A. Tuning forks, in fact, behave like
rather narrow bandpass filters. In the experiment with matched tuning forks, the filter was being driven by a signal
frequency corresponding to the peak in the filter's frequency response curve. Consequently, the filter produced a
great deal of energy at its output. In the experiment with mismatched tuning forks, the filter is being driven by a
signal that is remote from the peak in the filter's frequency response curve, producing a low amplitude output signal.
To summarize, resonance refers to the ability of one vibrating system to force another system into vibration.
Further, the amplitude of this forced vibration will be greater as the frequency of the driving force approaches the
natural vibrating frequency (resonance) of the system that is being forced into vibration.
3The increase in intensity that would occur as the tuning fork is placed in contact with a hard surface does not mean that additional energy is
created. The increase in intensity would be offset by a decrease in the duration of the tone, so the total amount of energy would not increase
relative to a freely vibrating tuning fork.

Cavity Resonators
An air-filled cavity exhibits frequency selective properties and should be considered a filter in precisely the way
that the tuning forks and wine glasses mentioned above are filters. The human vocal tract is an air-filled cavity that
behaves like a filter whose frequency response curve varies depending on the positions of the articulators. Tuning
forks and other simple filters have a single resonant frequency. (Note that we will be using the terms "natural
vibrating frequency" and "resonant frequency" interchangeably.) Cavity resonators, on the other hand, can have an

infinite number of resonant frequencies.
A simple but very important cavity resonator is the uniform tube. This is a tube whose cross-sectional area is
the same (uniform) at all points along its length. A simple water glass is an example of a uniform tube. The method
for determining the resonant frequency pattern for a uniform tube will vary depending on whether the tube is closed
at both ends, open at both ends, or closed at just one end. The configuration that is most directly applicable to
problems in speech and hearing is the uniform tube that is closed at one end and open at the other end. The ear canal,
for example, is approximately uniform in cross-sectional area and is closed medially by the ear drum and open
0.0
0.2
0.4
0.6
0.8
1.0
Gain
500 1500 2500 3500 4500
17.5 cm Uniform Tube
0.0
0.2
0.4
0.6
0.8
1.0
Gain
437.5 1312.5 2187.5 3062.5 3937.5
20 cm Uniform Tube
0 1000 2000 3000 4000 5000
0.0
0.2
0.4
0.6
0.8
1.0
Frequency (Hz)
Gain

583.3 1750.0 2916.7 4083.3 5225.0
15 cm Uniform Tube
Figure 3-23. Frequency response curves for three uniform tubes open at one end and closed at the
other. These kinds of tubes have an infinite number of resonances at odd multiples of the lowest resonance. As the figure shows, shortening the tube shifts all resonances to higher frequencies while
lengthening the tube shifts all resonances to lower frequencies.

laterally. Also, in certain configurations the vocal tract is approximately uniform in cross-sectional area and is
effectively closed from below by the vocal folds and open at the lips. The resonant frequencies for a uniform tube
closed at one end are determined by its length. The lowest resonant frequency (F1) for this kind of tube is given by:
F1 = c/4L, where: c = the speed of sound
L = the length of the tube
For example, for a 17.5 cm tube, F1 = c/4L = 35000/70 = 500 Hz. This tube will also have an infinite number of
higher frequency resonances at odd multiples of the lowest resonance:
F1 = F1 . 1 = 500 Hz
F2 = F1 · 3 = 1,500 Hz
F3 = F1 . 5 = 2,500 Hz
F4 = F1 · 7 = 3,500 Hz
The frequency response curve for this tube for frequencies below 4000 Hz is shown in the solid curve in Figure
3-23. Notice that the frequency response curve shows peaks at 500, 1500, 2500, and 3500 Hz, and valleys in
between these peaks. The frequency response curve, in fact, looks like a number of bandpass filters connected in
series with one another. It is important to appreciate that what we have calculated here is a series of natural vibrating
frequencies of a tube. What this means is that the tube will respond best to forced vibration if the tube is driven by
signals with frequencies at or near 500 Hz, 1500 Hz, 2500 Hz, and so on. Also, the resonant frequencies that were
just calculated should not be confused with harmonics. Harmonics are frequency components that are present in the
amplitude spectra of complex periodic sounds; resonant frequencies are peaks in the frequency response curve of

filters.
We next need to see what will happen to the resonant frequency pattern of the tube when the tube length
changes. If the tube is lengthened to 20 cm:
F1 = c/4L = 35,000/80 = 437.5 Hz
F2 = F1 · 3 = 1,312.5 Hz
F3 = F1 · 5 = 2,187.5 Hz
F4 = F1 · 7 = 3,062.5 Hz
It can be seen that lengthening the tube from 17.5 cm to 20 cm has the effect of shifting all of the resonant
frequencies downward (see Figure 3-23). Similarly, shortening the tube has the effect of shifting all of the resonant
frequencies upward. For example, the resonant frequency pattern for a 15 cm tube would be:
F1 = c/4L = 35,000/60 = 583.3 Hz
F2 = F1 · 3 = 1,750 Hz
F3 = F1 · 5 = 2,916.7 Hz
F4 = F1 · 7 = 4,083.3 Hz
The general rule is quite simple: all else being equal, long tubes have low resonant frequencies and short tubes
have high resonant frequencies. This can be demonstrated easily by blowing into bottles of various lengths. The
longer bottles will produce lower tones than shorter bottles. This effect is also demonstrated every time a water glass
is filled. The increase in the frequency of the sound that is produced as the glass is filled occurs because the
resonating cavity becomes shorter and shorter as more air is displaced by water. This simple rule will be quite
useful. For example, it can be applied directly to the differences that are observed in the acoustic properties of
speech produced by men, women, and children, who have vocal tracts that are quite different in length.

Resonant Frequencies and Formant Frequencies

The term "resonant frequency" refers to natural vibrating frequency or, equivalently, to a peak in a frequency
response curve. For reasons that are entirely historical, if the filter that is being described happens to be a human
vocal tract, the term formant frequency is generally used. So, one typically refers to the formant frequencies of the

vocal tract but to the resonant frequencies of a plastic tube, the body of a guitar, the diaphragm of a loudspeaker, or
most any other type of filter other than the vocal tract. This is unfortunate since it is possible to get the mistaken idea
that formant frequencies and resonant frequencies are different sorts of things. The two terms are, in fact, fully
synonymous.

The Decibel Scale

The final topic that we need to address in this chapter is the representation of signal amplitude using the decibel
scale. The decibel scale is a powerful and immensely flexible scale for representing the amplitude of a sound wave.
The scale can sometimes cause students difficulty because it differs from most other measurement scales in not just
one but two ways. Most of the measurement scales with which we are familiar are absolute and linear. The decibel
scale, however, is relative rather than absolute, and logarithmic rather than linear. Neither of these characteristics is
terribly complicated, but in combination they can make the decibel scale appear far more obscure than it is. We will
examine these features one at a time, and then see how they are put together in building the decibel scale.

Linear vs. Logarithmic Measurement Scales

Most measurement scales are linear. To say that a measurement scale is linear means that it is based on equal
additive distances. This is such a common feature of measurement scales that we do not give it much thought. For
example, on a centigrade (or Fahrenheit) scale for measuring temperature, going from a temperature of 90 o to a
temperature of 91 o involves adding one 1 o. One rather obvious consequence of this simple additivity rule is that the
difference in temperature between 10 o and 11 o is the same as the difference in temperature between 90 o and 91 o.
However, there are scales for which this additivity rule does not apply. One of the best known examples is the
Richter scale that is used for measuring seismic intensity. The difference in seismic intensity between Richter values
of 4.0 and 5.0, 5.0 and 6.0, 6.0 and 7.0 is not some constant amount of seismic intensity, but rather a constant

multiple. Specifically, a 7.0 on the Richter scale indicates an earthquake that is 10 times greater in intensity than an earthquake that measures 6.0 on the Richter scale. Similarly, an 8.0 on the Richter scale is 10 times greater in intensity than a 7.0. Whenever jumping from one scale value to the next involves multiplying by a constant rather than adding a constant, the scale is called logarithmic. (The multiplicative constant need not be 10. See Box 3-2 for an example of a logarithmic scale – an octave progression – that uses 2 as the constant.) Another way of making the same point is to note that the values along the Richter scale are exponents rather than ordinary numbers; for example, a Richter value of 6 indicates a seismic intensity of 10^6, a Richter value of 7 indicates a seismic intensity of 10^7, etc. The Richter values can, of course, just as well be referred to as powers or logarithms since both of these terms are synonyms for exponent. The decibel scale is an example of a logarithmic scale, meaning that it is based on equal multiples rather than equal additive distances.

Absolute vs. Relative Measurement Scales

A simple example of a relative measurement scale is the Mach scale that is used by rocket scientists to measure speed. The Mach scale measures speed not in absolute terms but in relation to the speed of sound. For example, a missile at Mach 2.0 is traveling at twice the speed of sound, while a missile at Mach 0.9 is traveling at 90% of the speed of sound. So, the Mach scale does not represent a measured speed (S_m) in absolute terms, but rather, represents a measured speed in relation to a reference speed (S_m/S_r). The reference that is used for the Mach scale is the speed of sound, so a measured absolute speed can be converted to a relative speed on the Mach scale by simple division. For example, taking 783 mph as the speed of sound, 1,200 mph = 1200/783 = Mach 1.53. The decibel scale also exploits this relative measurement scheme. The decibel scale does not represent a measured intensity (I_m) in absolute terms, but rather, represents the ratio of a measured intensity to a reference intensity (I_m/I_r).

The decibel scale is trickier than the Mach scale in one important respect. For the Mach scale, the reference is

always the speed of sound, but for the decibel scale, many different references can be used. In explaining how the
decibel scale works, we will begin with the commonly used intensity reference of 10 -12 w/m2 (watts per square
meter), which is approximately the intensity that is required for an average normal hearing listener to barely detect a
1,000 Hz pure tone. So, for our initial pass through the decibel scale, 10 -12 w/m2 will serve as I r , and will perform the
same function that the speed of sound does for the Mach scale. Table 3-1 lists several sounds that cover a very broad
range of intensities. The second column shows the measured intensities of those sounds, and the third column shows
the ratio of those intensities to our reference intensity. Whispered speech, for example, measures approximately 10 -8

w/m2, which is 10,000 times more intense than the reference intensity (10 -8 /10 -12 = 10 4 = 10,000). The main point to
be made about column 3 is that the ratios become very large very soon. Even a moderately intense sound like
conversational speech is 1,000,000 times more intense than the reference intensity. The awkwardness of dealing
with these very large ratios has a very simple solution. Column 4 shows the ratios written in exponential notation,
and column 5 simplifies the situation even further by recording the exponent only. The term exponent and the term
logarithm are synonymous, so the measurement scheme that is expressed by the numbers in column 5 can be
summarized as follows: (1) divide a measured intensity by a reference intensity (in this case, 10 -12 w/m2), (2) take the
logarithm of this ratio (i.e., write the number in exponential notation and keep the exponent only). This method, in
fact, is a completely legitimate way to represent signal intensity. The unit of measure is called the bel, after A.G.
Bell, and the formula is:
bel = log10 I m/I r , where: I m = a measured intensity
I r = a reference intensity

_____
_____
_____
_____

Table 3-1. Sound intensities and intensity ratios showing how the decibel scale is created. Column 2 shows the
measured intensities (Im) of several sounds. Column 3 shows the ratio of these intensities to a reference intensity of
10 -12 w/m2 . Column 4 shows the ratio written in exponential notation while column 5 shows the exponent only. The
last column shows the intensity ratio expressed in decibels, which is simply the logarithm of the intensity ratio
multiplied by 10.

Measured Sound Intensity (I m)	Ratio Ratio in (Im/Ir)	Exponent Exp. Not. (log 10)	Decibel (10 x log 10)
Threshold @ 1 kHz	10 -12 w/m2	1	10 0 0 0
Whisper	10 -8 w/m2	10,000	10 4 4 40
Conversational Speech	10 -6 w/m2	1,000,000	10 6 6 60
City Traffic	10 -4 w/m2	100,000,000	10 8 8 80
Rock & Roll	10 -2 w/m2	10,000,000,000	10 10 10 100
Jet Engine	10 0 w/m2	1,000,000,000,000	10 12 12 120

Legitimate or not, the bel finds its sole application in textbooks attempting to explain the decibel. For reasons that
are purely historical, the log10 of the intensity ratio is multiplied by 10, changing bel into the decibel (dB). As shown
in the last column of Table 3-1, this has the very simple effect of turning 4 bels into 40 decibels, 8 bels into 80
decibels, etc. The formula for the decibel, then, is:

$$dB\ IL = 10 \log_{10} I_m / I_r$$

, where:

I_m = a measured intensity

I_r = a reference intensity

The designation "IL" stands for intensity level, and it indicates that the underlying measurements are of sound
intensity and not sound pressure. As will be seen below, a different version of this formula is needed if sound

pressure measurements are used. The multiplication by 10 in the dB IL formula is a simple operation, but it can

sometimes have the unfortunate effect of making the formula appear more obscure that it is. The decibel values that
are calculated, however, should be readily interpretable. For example, 30 dB IL means 3 factors of 10 more intense
than I r , 60 dB IL means 6 factors of 10 more intense than I r , and 90 dB IL means 9 factors of 10 more intense than I r .

Deriving a Pressure Version of the dB Formula

In a simple world, we would be finished with the decibel scale. The problem is that the formula is based on
measurements of sound intensity, but as a purely practical matter sound intensity is difficult to measure. Sound
pressure, on the other hand, is quite easy to measure. An ordinary microphone, for example, is a pressure sensitive
device. The problem, then, is that the decibel is defined in terms of intensity measurements, but the measurements
that are actually used will nearly always be measures of sound pressure. This problem can be addressed since there
is a predictable relationship between intensity (I) and pressure (E): intensity is proportional to pressure squared:

I o⊂ E 2

Knowing this relationship allows us to create a completely equivalent version of the decibel formula that will work
when sound pressure measurements are used instead of sound intensity measurements. All we need to do is
substitute squared pressure measurements in place of the intensity measurements:

dB IL = 10 log10 I m/I r (intensity version of formula)

dB SPL= 10 log10 E 2m/E 2 (pressure version of formula)

The designation "SPL" stands for sound pressure level, and it indicates that measures of sound pressure have been
used and not measures of sound intensity. Although the dB SPL formula shown here will work fine, it will almost
never be seen in this form. The reason is that the formula is algebraically rearranged so that the squaring operation is
not needed. The algebra is shown below:

(1) dB IL = 10 log10 I m/I r (the intensity version of the formula)

(2) dB SPL = 10 log10 E 2m/E 2 (measures of E 2 replace measures of I because I o⊂ E 2)

(3) dB SPL = 10 log10 (E m/E r) 2 (a 2

/b 2 = (a/b) 2)
(4) dB SPL = 10 · 2 log10 E m/E r (this is the only tricky step: log a b = b log a)
(5) dB SPL = 20 log10 E m/E r (2 · 10 = 20)
With the possible exception of the fourth step,4 the algebra is straightforward, but the details of the derivation
are less important than the following general points:
1. The decibel formula is defined in terms of intensity ratios. The basic formula is;
dB IL = 10 log10 I m/I r .
2. While sound intensity is difficult to measure, sound pressure is easy to measure. It is therefore necessary to
derive a version of the decibel formula that works when measures of sound pressure are used instead of sound
intensity.
4 Step 4 is the only tricky part of derivation. The reason it works is that squaring a number and then taking a log is the same as taking
the log first, and then multiplying the log by 2. For example, note that the two calculations below produce the same result:
log 10 100 2 = log 10 10,000 = 4 (square first, then take the log)
log 10 100 2 = (log 10 100) x 2 = 2 x 2 = 4 (take the log, then multiply by 2)

3. The derivation of the pressure version of the formula is based entirely on the fact that intensity is proportional to
pressure squared (I o⊂ E 2). This allows measures of E 2 to replace measures of I, turning: dB IL = 10 log10 I m/I r into
dB SPL = 10 log10 E 2m/E r2 . A few algebra tricks are applied to turn this formula into the more aesthetically pleasing
final version: dB SPL = 20 log10 E m/E r .
4. The two versions of the formula are fully equivalent to one another (see Box 3-3).
This last point about the equivalence of the intensity and sound pressure versions of the formula is explained in
some detail in Box 3-3, but the basic point is quite simple. The pressure version of the dB formula was derived from
the intensity version of the formula through algebraic manipulations (based on this relationship: I o⊂ E 2). The whole

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Box 3-2
HARMONICS, OCTAVES, LINEAR SCALES, AND LOGARITHMIC SCALES

As we will see when the decibel scale is introduced, there is an important distinction to be made between
linear scales, which are quite common, and logarithmic scales, which are less common but quite important.
This distinction can be illustrated by examining the difference between a harmonic progression and an octave
progression. Notice that in a harmonic progression, the spacing between the harmonics is always the same; that
is, the difference between H1 and H 2 is the same as the difference between H2 and H 3, and so on. This is because
increases in frequency between one harmonic and the next involve adding a constant, with the constant being
the fundamental frequency. For example:
H 1 500
H 2 1000 (add 500)
H 3 1500 (add 500)
H 4 2000 (add 500)
. .
. .
. .
To get from one scale value to another on an octave progression involves multiplying by a constant rather
than adding a constant. For example, an octave progression starting at 500 Hz looks like this:
O 1 500
O 2 1000 (multiply by 2)
O 3 2000 (multiply by 2)
O 4 4000 (multiply by 2)
. .
. .
. .
As a result of the fact that we are multiplying by a constant rather than adding a constant, the spacing is no
longer even (i.e., the spacing between O 1 and O 2 is 500 Hz, the spacing between O2 and O 3 is 1000 Hz, and so
on). The point to be made of this is that there are two fundamentally different kinds of scales: (1) scales like
harmonic progressions that are created by adding a constant, which are by far the more common, and (2) scales
like octave progressions that are created by multiplying by a constant. Scales that are created by adding a

constant are called linear scales, while scales that are created by multiplying by a constant are called
logarithmic scales. Note that for an octave progression, the multiplier happens to be 2, meaning that progressing
from one frequency to an octave above that frequency involves multiplication by 2. However, a logarithmic
scale can be built using any multiplier. We will return to the distinction between linear and logarithmic scales
when we talk about the decibel scale, and there we will see that a logarithmic scale is built around multiplication
by a constant value of 10 rather than 2.

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point of algebra, of course, is to keep the expression on the left equal to the expression on the right. The simple and
useful point that emerges from this is this: If an intensity meter shows that a given sound measures 60 dB IL, for
example, a pressure meter will show that the same sound measures exactly 60 dB SPL. (This may seem
counterintuitive due to the differences in the formulas, but see Box 3-3 for the explanation.) The equivalence of the
two versions of the dB formula greatly simplifies the interpretation of sound levels that are expressed in decibels.

References

The reference that is used for the Mach scale is always the speed of sound. One of the virtues of the decibel
scale is that any reference can be used as long as it is clearly specified. The only reference that has been mentioned
so far is 10^{-12} w/m2
, which is roughly the audibility threshold for a 1,000 Hz pure tone. This is a standard reference intensity, and unless otherwise stated it should be assumed that this is used when a signal level is reported in dB IL.

The standard reference that is used for dB SPL is 20 μPa, so when a signal level is reported in dB SPL it should be
assumed that this reference is used unless otherwise stated.5

Many references besides these two standard references can be used. For example, suppose that a speech signal

is presented to a listener at an average level of 3500 μPa in the presence of a noise signal whose average sound
pressure is 1400 μPa. The speech-to-noise ratio (S/N) can be represented on a decibel scale, using the level of the
speech as E m and the level of the noise as E r :

dB s/n = 20 log10 E m/ E r
= 20 log10 3500/1400
= 20 log10 2.5
= 20 (0.39794)
= 7.96 dB

To take one more example, assume that a voice patient prior to treatment produces sustained vowels that
average 2300 μPa. Following treatment the average sound pressures increase to 8890 μPa. The improvement in
sound pressure (post-treatment relative to pre-treatment) can be represented on a decibel scale:

dB Improvement = 20 log10 E post/E pre
= 20 log10 8890/2300
= 20 log10 (3.86522)
= 20 (0.58717)
= 11.74 dB

A final example can be used to make the point that the decibel scale can be used to represent intensity ratios for
any type of energy, not just sound. Bright sunlight has a luminance measuring 100,000 cd/m2 (candela per square
meter). Light from a barely visible star, on the other hand, has a luminance measuring 0.0001 cd/m2. We can now
ask how much more luminous bright sunlight is in relation to barely visible star light, and the dB scale can be used
to represent this value. Since the underlying physical quanities here are measures of electromagnetic intensity, we
want the intensity version of the formula rather than the pressure version.

dB = 10 log10 I sunlight/I starlight
= 10 log10 100000/0.0001
= 10 log10 10 5/10 -4
= 10 log10 10 9 (division is done by subtracting exponents: 5 – (-4) = 9)
= 10 (9)
= 90 dB

5The standard pressure reference for dB SPL is sometimes given as 0.0002 dynes/cm2 rather than 20 μPa. These two sound pressures are

identical, however, in exactly the same sense that 4 quarts and 1 gallon are identical. Likewise, the standard reference for dB IL is often given as
10 -16 w/cm2 instead of 10 -12 w/m2
. These two intensities are also identical.

The fact that we are measuring light rather than sound makes no difference: a decibel is 10 log10 I m/I r (or,
equivalently, 20 log10 E m/E r), regardless of whether the energy comes from sound, light, electrical current, or any
other type of energy.

dB Hearing Level (dB HL)

The dB Hearing Level (dB HL) scale was developed specifically for testing hearing sensitivity for pure tones
of different frequencies. The sound-level dials on clinical audiometers,6 for example, are calibrated in dB HL rather
than dB SPL. To understand the motivation for the dB HL scale examine Figure 3-24, which shows the sound level (in
dB SPL) required for the average, normal-hearing listener to barely detect pure tones at frequencies between 125 and
8000 Hz. This is called the audibility curve and the simple but very important point to notice about this graph is
that the curve is not a flat line; that is, the ear is clearly more sensitive at some frequencies than others. The
differences in sensitivity are quite large in some cases. For example, the average normal-hearing listener will barely
detect a 1000 Hz pure tone at 7 dB SPL, but at 125 Hz the sound level needs to be cranked all the way up to 45 dB SPL,
an increase in intensity of nearly 4000:1. Now suppose we were to test pure-tone sensitivity using an audiometer that
is calibrated in dB SPL. Imagine that a listener barely detects a 1000 Hz pure tone at 25 dB SPL. Does this listener have
a hearing loss, and if so how large? The only way to answer this question is to consult the data in Figure 3-24, which
shows that the threshold of audibility for the average normal hearing listener at 1000 Hz is 7 dB SPL. This means that
the hypothetical listener in this example has a hearing loss of 25-7 = 18 dB. Suppose further that the same listener
detects a 250 Hz tone at 20 dBSPL. The table in Figure 3-24 shows that normal hearing sensitivity at 250 Hz is 25.5

dB SPL, meaning that the listener has slightly better than normal hearing at this frequency. As a final example,
imagine that this listener barely detects a 500 Hz tone at 30 dB SPL. Since the table shows that normal hearing
sensitivity at 500 Hz is 11.5 dB SPL, the listener has a hearing loss of 30.0-11.5 = 18.5 dB. The simple point to be
made about these examples is that, with an audiometer dial that is calibrated in dB SPL, it is not possible to determine
whether a listener has a hearing loss, or to measure the size of that loss, without doing some arithmetic involving the
normative data in Figure 3-24. The dB HL scale, however, provides a simple solution to this problem that avoids this
arithmetic entirely. The solution involves calibrating the audiometer in such a way that, when the level dial is set to
0 dB HL, sound level is set to the threshold of audibility for the average normal-hearing listener for that signal
frequency. For example, when the level dial is set to 0 dB HL at 125 Hz the level of tone will be 45 dB SPL – the
threshold of audibility for the average normal hearing listener at this frequency. Now if a listener barely detects the
125 Hz tone at 0 dB HL, no arithmetic is needed; the listener has normal hearing at this frequency. Further, if the
listener barely detects this 125 Hz tone at 40 dB HL, for example, the listener must have a 40 dB loss at this frequency
– and again it is not necessary to consult the data in Figure 3-24. Similarly, when the level dial is set to 0 dB HL at
250 Hz the level of the tone will be 25.5 dB SPL, which is the audibility threshold at 250 Hz. If this tone is barely
detected at 0 dB HL, the listener has normal hearing at this frequency. However, if the tone is not heard until the dial
is increased to 50 dB dB HL, for example, the listener has a 50 dB hearing loss at this frequency. The same system is
used for all signal frequencies: in all cases, the 0 dB HL reference is not a fixed number as it is for dB SPL (a constant
value of 20 μPa, no matter what the signal frequency is) or dB IL (a constant value of 10 -12 watts/m2, again
independent of signal frequency), but rather a family of numbers. In each case the reference for the dB HL scale is the
threshold of audibility for an average, normal-hearing listener at a particular signal frequency. What this means is

that values in dB HL are a fixed distance above the audibility curve, although they may be very different levels in
dB SPL. For illustration, Figure 3-25 shows the audibility curve (the filled symbols) and, above that in the unfilled
symbols, a collection of values that all measure 30 dB HL. Although the sound levels on the 30 dB HL curve vary
considerably in dB SPL (i.e. measured using 20 μPa as the reference), every data point on this curve is a constant 3
factors of 10, or 30 dB, above the audibility curve. The value of 30 dB in this figure is just an example. All values in
dB HL and dB SPL are interpreted in the same way: 50 dB SPL means that the signal being measured is 100,000 times
(i.e., 5 factors of 10) more intense than the fixed reference of 20 μPa, independent of frequency; 50 dB HL, on the
other hand, means that the signal being measured is 100,000 times (again, 5 factors of 10) more intense than a tone
that is barely audible to a normal-hearing listener at that signal frequency. Similarly, 20 dB SPL means that the signal
is 20 dB (2 factors of 10) more intense than the fixed reference of 20 μPa, while 20 dB HL means that the signal is 20
dB (again, 2 factors of 10) above the audibility curve.
6A clinical audiometer is an instrument with, among other things, one dial (for each ear) that controls pure-tone frequency and another dial
that controls the intensity of the tone. The listener is asked to raise a hand when the tone is barely audible.

Summary
The decibel is a powerful scale for representing signal amplitude. The scale has two important properties: (1)
similar to the Mach scale, it represents signal level not in absolute terms but as a measured level divided by a
reference level; and (2) like the Richter scale, the dB scale is logarithmic rather than linear, meaning that it is based
on equal multiplicative distances rather than equal additive distances. While the decibel is defined in terms of
intensity ratios, for practical reasons, measures of sound pressure are far more common than measures of sound
intensity. Consequently, a version of the decibel formula was derived that makes use of pressure ratios rather than

intensity ratios. The derivation was based on the fact that intensity is proportional to pressure squared. The two
versions of the decibel formula (dB IL = 10 log 10 I m/I r and dB SPL = 20 log 10 E m/E r) are fully equivalent, meaning that
if a sound measures 60 dB IL that same sound will measure 60 dB SPL. Unlike the Mach scale, which always uses the
speed of sound as a reference, any number of references can be used with the decibel scale. The standard reference
for the dB IL scale is 10 -12 w/m2 and the standard reference for the dB SPL scale is 20 μPa. However, any level can be
used as a reference as long as it is specified. The dB HL scale, widely used in audiological assessment, was developed
specifically for measuring sensitivity to pure tones of difference frequencies. The reference that is used for the dB HL
scale is the threshold of audibility at a particular signal frequency for the average, normal-hearing listener. Sound
levels in dB SPL and dB HL are interpreted quite differently. For example, a pure tone measuring 40 dB SPL is 4 factors
of 10 (i.e., 40 dB) greater than the fixed SPL reference of 20 μPa, while a pure tone measuring 40 dB HL is 4 factors
of 10 (again, 40 dB) greater than a tone of that same frequency that is barely audible to an average, normal-hearing
listener.

Frequency	Threshold
125	45.0
250	25.5
500	11.5
750	8.0
1000	7.0
1500	6.5
2000	9.0
3000	10.0
4000	9.5
6000	15.5
8000	13.0

Figure 3-24. The threshold of audibility for the average, normal-hearing listener for pure tones varying between
125 and 8000 Hz. The audibility threshold is the sound level in dBSPL that is required for a listener to barely detect

a tone. Values on this curve are shown in the table to the right. The most important point to note about this graph
is that the curve is not flat, meaning that the ear is more sensitive at some frequencies than others. In particular,
the ear is more sensitive in a range of mid-frequencies between about 1000 and 4000 Hz than it is at lower and
higher frequencies. The complex shape of this curve provides the underlying motivation for the dB HL scale. See
text for details.
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Figure 3-25. The lower function is the audibility curve – the sound level in dB SPL that is required for an average
normal hearing listener to barely detect pure tones of different frequencies. The upper function shows sound levels for
a set of tones that all measure 30 dB HL. These tones vary quite a bit in dB SPL (i.e., relative to the constant value of 20
μPa) but in all cases the tones are a constant 3 factors of 10 in intensity (i.e., 30 dB) above the audibility curve.
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Box 3-3
THE EQUIVALENCE OF THE INTENSITY AND PRESSURE
VERSIONS OF THE DECIBEL FORMULA
One fact about the two versions of the dB formula that is not always well understood is that the dB IL and
dB SPL formulas are fully equivalent. By "fully equivalent" we mean the following: suppose that a sound intensity
meter is used to measure the level of some sound, and we find that this sound is 1,000 times more intense than
the standard intensity reference of 10 -12 w/m2. The sound would then measure 30 dB IL (10 log10 1,000 = 10 (3) =
30 dB IL). Now suppose that we put the sound intensity meter away and use a sound pressure meter to measure
the same sound. You might think that the sound would measure 60 dB SPL since now we are multiplying by 20
instead of 10, but the trick is that the ratio is no longer 1,000. Recall that intensity is proportional to pressure

squared, which means that pressure is proportional to the square root of intensity. This means that if the intensity
ratio is 1,000, the pressure ratio must be the square root of 1,000, or 31.6. So, the formula now becomes 20 log
31.6 = 20 (1.5) = 30 dB SPL, which is exactly what we obtained originally. It will always work out this way: if a
sound measures 50 dB IL, that same sound will measure 50 dB SPL.
Table 3-2 might help to make this more clear. The first column shows an intensity ratio, the second column
shows the corresponding pressure ratio (this is always the square root of the intensity ratio), the third column
shows the dB IL value (10 log of the intensity ratio), and the fourth column shows dB SPL value (20 log of the
pressure ratio). As you can see, they are always the same.

Table 3-2. Intensity ratios, equivalent pressure ratios, dB IL values and dB SPL values showing the equivalence of the intensity and pressure versions of the dB formula.

Intensity Ratio	Pressure Ratio	dB IL (10 log10 I m/I r)	dB SPL (20 log10 E m/E r)
10	3.16	10.00	10.00
20	4.47	13.01	13.01
40	6.32	16.02	16.02
50	7.07	16.99	16.99
60	7.75	17.78	17.78
70	8.37	18.45	18.45
80	8.94	19.03	19.03
90	9.49	19.54	19.54
100	10.00	20.00	20.00
200	14.14	23.01	23.01
300	17.32	24.77	24.77
400	20.00	26.02	26.02
500	22.36	26.99	26.99
1000	31.62	30.00	30.00

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Study Questions: Physical Acoustics

1. Explain the basic processes that are involved in the propagation of a sound wave.
2. Draw time- and frequency-domain representations of simple periodic, complex periodic, complex aperiodic, and transient sounds.
3. Draw time- and frequency-domain representations of two complex periodic sounds with different fundamental frequencies.
4. Draw time-domain representations of two simple periodic sounds with the same frequency and phase, but different amplitudes.
5. Draw time-domain representations of two simple periodic sounds with the same frequency and different amplitudes but different phases.
6. Draw amplitude spectra of two sounds with the same fundamental frequencies but different spectrum envelopes.
7. Draw amplitude spectra of two sounds with different fundamental frequencies but similar spectrum envelopes.
8. Calculate signal frequencies for sinusoids with the following values:

a. period = 0.34 s
b. period = 2 s
c. period = 10 ms
d. period = 2 ms
e. wavelength = 20 cm
f. wavelength = 100 cm

Answers:

a. f = 1/0.34 = 2.94 Hz
b. f = 1/2 = 0.5 Hz
c. f = 1/0.01 = 100 Hz
d. f = 1/.002 = 500 Hz
e. f = c/WL (speed of sound/wavelength) = 35000/20 = 1750 Hz
f. f = c/WL (speed of sound/wavelength) = 35000/100 = 350 Hz

9. Calculate the three lowest resonant frequencies of the following uniform tubes that are closed at one end and open at the other end:

a. 10 cm
b. 30 cm
c. 40 cm

Answers:

a. wavelength of lowest resonance = 40 cm (10 x 4)
f = 35000/40 = 875
R1 = 875 (R1 = frequency of resonance number 1)
R2 = 2625
R3 = 4375
b. wavelength of lowest resonance = 120 cm (30 x 4)
f = 35000/120 = 291.7

R1 = 291.7
R2 = 875.0
R3 = 1458.3
c. wavelength of lowest resonance = 160 cm (40 x 4)
f = 35000/160 = 218.75
R1 = 218.75
R2 = 656.25
R3 = 1093.75
10. Show what the frequency-response curves look like for the tubes in the problem above.
11. A complex periodic signal has a fundamental period of 4 msec. What is the fundamental frequency of the signal? At
what frequencies would we expect to find energy?
12. How are the terms octave and harmonic different?
13. Give examples of the following kinds of graphs, being sure to label both axes:
a. amplitude spectrum
b. phase spectrum
c. frequency-response curve
d. time-domain representation
14. Give a brief explanation of the basic idea behind Fourier analysis. What is the input to Fourier analysis and what kind
of output(s) does it produce?
15. Draw and label frequency-response curves for low-pass, high-pass, and band-pass filters.
16. What parameters control the frequency of vibration of a spring and mass system?
17. Draw the time domain representation of one cycle of a sinusoid as variations in instantaneous air pressure over time
and one cycle of that same sinusoid as variations in instantaneous velocity over time.
18. How, if at all, are the terms resonant frequency and harmonic different?
19. How, if at all, are the terms resonant frequency and formant different?
20. A harmonic is a peak in: (a) a frequency response curve, (b) an amplitude spectrum, or (c) either a frequency response
curve or an amplitude spectrum.

21. A resonance is a peak in: (a) a frequency response curve, (b) an amplitude spectrum, or (c) either a frequency response
curve or an amplitude spectrum.
22. A formant is a peak in: (a) a frequency response curve, (b) an amplitude spectrum, or (c) either a frequency response
curve or an amplitude spectrum.
23. A frequency response curve describes a _____.
24. An amplitude spectrum describes a _____.

Frequency Response Problems

Answers to Frequency Response Problems

Decibel Study Questions
1. What reference is used for the dB IL scale?
2. What reference is used for the dB SPL scale?
3. What reference is used for the dB HL scale?
4. What reference is used for the dB SL scale?
5. A listener barely detects a 125 Hz pure tone at 55 dB SPL. Does this listener have a hearing loss at 125 Hz, and if
so, what is the size of the hearing loss?
6. A listener barely detects a 1,000 Hz pure tone at 55 dB SPL. Does this listener have a hearing loss at 1,000 Hz,
and if so, what is the size of the hearing loss?
7. A listener barely detects a 125 Hz pure tone at 55 dB HL. Does this listener have a hearing loss at 125 Hz, and if
so, what is the size of the hearing loss?
8. A listener barely detects a 1,000 Hz pure tone at 55 dB HL. Does this listener have a hearing loss at 1,000 Hz,
and if so, what is the size of the hearing loss?
9. 60 dB SPL at 1,000 Hz means _____ more intense than
_____.
10. 60 dB IL at 1,000 Hz means _____ more intense than
_____.
11. 60 dB HL at 1,000 Hz means _____ more intense than
_____.
12. The reference that is used for the dB SPL scale is:
a. a number
b. a sentence

13. If the answer to the question above is a number, give the number; if it's a sentence, give the sentence.
14. The reference that is used for the dB HL scale is:
a. a number
b. a sentence
15. If the answer to the question above is a number, give the number; if it's a sentence, give the sentence.
16. A specific individual has a 70 dB hearing loss in the left ear at 1,000 Hz. A 90 dB HL, 1,000 Hz tone that is
presented to this listener's left ear would measure _____ dB SL.
17. A sound measures 42 dB IL. On the dB SPL scale, that same sound will measure:
a. 84 dB SPL because with the dB SPL formula we are now are multiplying the ratio by 20 instead of 10.
b. 42 dB SPL because the two versions of the formula are equivalent
18. A sound measures 60 dB IL. (a) The measured intensity (I M) must therefore be _____ times
greater than the reference intensity (I R). (b) What would the pressure ratio (E M/E R) be for this same sound? (c)
Do the arithmetic to show what this sound would measure in dB SPL.

19. A sound measures 40 dB IL. (a) The measured intensity (I M) must therefore be _____ times
greater than the reference intensity (I R). (b) What would the pressure ratio (E M/E R) be for this same sound? (c)
Do the arithmetic to show what this sound would measure in dB SPL.
20. On the graph below, put a mark at: (a) 3,000 Hz, 20 dB SPL, and (b) 3,000 Hz, 20 dB HL (the
grid lines on the y axis are spaced at 2 dB intervals).

Frequency in Hz	Threshold in dB SPL
125	45.0
250	25.5
500	11.5
750	8.0
1000	7.0
1500	6.5
2000	9.0
3000	10.0
4000	9.5
6000	15.5

8000 13.0
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Answers to Decibel Study Questions
1. 10 -12 watts/m2
2. 20 µPa (or, equivalently, 0.0002 dynes/cm2)
3. The threshold of audibility for an average, normal-hearing listener at a particular signal frequency.
4. 3. The threshold of audibility for a particular listener at a particular signal frequency.
5. Consulting the attached figure and table showing the audibility curve for average, normal-hearing listeners, we
find that the threshold of audibility at 125 Hz is 45 dB SPL. A listener who barely detected a 125 Hz tone at 55
dB SPL would therefore have hearing loss of 55-45=10 dB; that is, the hearing sensitivity of this listener would be
10 dB worse than normal.
6. Consulting the attached figure and table showing the audibility curve for average, normal-hearing listeners, we
find that the threshold of audibility at 1,000 Hz is 7 dB SPL. A listener who barely detected a 1,000 Hz tone at
55 dB SPL would therefore have a hearing loss of 55-7=48 dB; that is, the hearing sensitivity of this listener
would be 48 dB worse than normal.
7. The reference for dB HL is the audibility threshold, so this listener would have a 55 dB hearing loss at 125 Hz.
There is no need to consult the table.
8. The reference for dB HL is the audibility threshold, so this listener would have a 55 dB hearing loss at 1,000 Hz.
There is no need to consult the table.
9. 6 factors of 10 (i.e., 1,000,000 times) more intense than 20 µPa)
10. 6 factors of 10 (i.e., 1,000,000 times) more intense than 10 -12 watts/m2
11. 6 factors of 10 (i.e., 1,000,000 times) more intense than a 1,000 Hz tone that is barely audible to an average,
normal-hearing listener.
12. a number
13. 20 µPa
14. a sentence
15. The threshold of audibility for an average, normal-hearing listener at a particular signal frequency.
16. 20 dB SL. The reference for the dB SL (SL=sensation level) is the threshold of audibility for a specific listener. So,

what we want to know here very simply is where this 90 dB HL tone is in relation to this particular listener's
threshold. This listener has a 70 dB hearing loss at this frequency, so the 90 dBHL tone, which would be 90 dB
above a normal-hearing listener's threshold, is only 20 dB above this particular listener's threshold.
17. 42 dB SPL: The pressure version of the formula was derived from the intensity version through algebraic
manipulations, so they have to be equivalent to one another. The next problem was designed to illustrate how
this can be the case.
18. (a) 1,000,000 times (6 factors of 10) more intense than I R. (b) If the intensity ratio is 1,000,000, the pressure
ratio has to be the square root of 1,000,000, which is 1,000. (c) dB SPL = 20 log 1,000 = 20 . 3 = 60 dB SPL. This is
exactly what we got for the same sound measured in dB IL. It will always be the same. If a sound measures 60
dB IL, that same sound will measure 60 dB SPL.
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19. (a) 10,000 times (4 factors of 10) more intense than I R. (b) If the intensity ratio is 10,000, the pressure ratio has
to be the square root of 10,000, which is 100. (c) dB SPL = 20 log 100 = 20 . 2 = 40 dB SPL. This is exactly what
we got for the same sound measured in dB IL. It will always be the same. If a sound measures 40 dB IL, that same
sound will measure 40 dB SPL.
20. See below. The lower of the two marks is 20 dB (2 factors of 10) above the constant reference line of 20 μPa.
The higher of the two marks is 20 dB (also 2 factors of 10) above the curvey line, which is the threshold of
audibility for the average normal-hearing listener.
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A Tutorial on Digital Sound Synthesis Techniques
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A Tutorial on Digital Sound Synthesis Techniques

Introduction

Progress in electronics and computer technology has led to an ever-increasing utilization of digital techniques for musical sound production. Some of these are the digital equivalents of techniques employed in analog synthesizers and in other fields of electrical engineering. Other techniques have been specifically developed for digital music devices and are peculiar to these.

This paper introduces the fundamentals of the main digital synthesis techniques. Mathematical developments have been restricted in the exposition and can be found in the papers listed in the references. To simplify the discussion, whenever possible, the techniques are presented with reference to continuous signals.

Sound synthesis is a procedure used to produce a sound without the help of acoustic instruments. In

digital synthesis, a sound is represented by a sequence of numbers (samples). Hence, a digital synthesis technique consists of a computing procedure or mathematical formula, which computes each sample value.

Normally, the synthesis formula depends on some values, that is, parameters. Frequency and amplitude are examples of such parameters. Parameters can be constant or slowly time variant during the sound. Time-variant parameters are also called control functions.

Synthesis techniques can be classified as (1) generation techniques (Fig. la), which directly produce the signal from given data, and (2) transformation techniques (Fig. Ib), which can be divided into two stages, the generation of one or more simple signals and their modification. Often, more or less elaborate combinations of these techniques are employed.

Fixed-Waveform Synthesis

In many musical sounds, pitch is a characteristic to which we are quite sensitive. In examining the temporal shape of pitched sounds, we see a periodic repetition of the waveform without great variations. The simplest synthesis method attempts to reproduce this characteristic, generating a periodic signal through continuous repetition of the waveform. This method is called fixed-waveform synthesis.

The technique is carried out by a module called an oscillator (Fig. 2), which repeats the waveform with a specified amplitude and frequency. In certain cases, the waveform is characteristic of the oscillator and cannot be changed. But often it can be chosen in a predetermined set of options or given explicitly when required.

Usually, in digital synthesis the waveform value at a particular instant is not computed anew for each sample. Rather, a table, containing the period values computed in equally spaced points, is built beforehand. Obviously, the more numerous the

points in the table, the better the approximation will be. To produce a sample, the oscillator requires the waveform value at that precise instant. It cyclically searches the table to get the point nearest to the required one. Sometimes a finer precision is achieved by interpolation between two adjacent points.

The distance in the table between two samples read at subsequent instants is called the sampling_increment. The sampling _increment is proportional to the frequency f of the generated signal according to the following formula (Mathews 1969):

N
samplingincrement = SRf,

where N is the table length and SR the sampling rate.

In the oscillator, the frequency is usually speci-

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Fig. 1. Classification of synthesis techniques. Generation techniques (a) and transformation techniques (b).

Fig. 2. Fixed-waveform synthesis oscillator.

(a) Parameters
'[M SoundGeneration S
signal
Complex

(b) Parameters sound
I signal
Generation Transformation
Simple
signals
A(t) f(t)

s(t)
fled as a sampling- increment and the algorithm that realizes it is as follows:
signal [t] := amplitude * table [phase],
(Relation 1)
and
phase := mod(n, phase + samplingincrement),
(Relation 2)
where
Table contains one period of the waveform;
Phase is the theoretical position in the table of the sample to be extracted at the instant; and
Amplitude is the signal amplitude.
Relation 2 computes the phase value in the subsequent instant, approximating the frequency integration by a summation. The modulus operation keeps the phase inside the table length n.
It is noteworthy that the signal generated in this way is an approximation of the desired one (Mailliard 1976). The approximation depends on the table length, the interpolation method, and the signal frequency. For a sufficiently long table, it is fully satisfactory.
The results of fixed-waveform synthesis are of poor musical quality, as the sound does not present any variation along its duration. This technique can be changed by allowing the amplitude to vary in time. In real sounds, the amplitude is rarely constant: it starts from zero, reaches a maximum after a certain time (attack), remains nearly constant (steady state) and, after a certain evolution, it returns to zero (decay). This sequence of amplitude behavior is called the envelope. Thus, when the amplitude varies according to a control function, we have fixed-waveform synthesis with an amplitude envelope.
The envelope can be generated in many ways.
In software-based synthesis, the most frequent method uses an oscillator module, seen previously, using a very low frequency equal to the inverse of

the duration. In this case, it performs a single cycle and its waveform corresponds to the amplitude envelope.

By carefully analyzing natural periodic sounds, it has been shown that even the most stable ones contain small frequency fluctuations. These improve the sound quality and avoid unpleasant beatings when more sounds are present at the same time.

The fixed-waveform technique can also be modified so that the oscillator frequency can slowly vary around a value. This enables the production of a tremolo and, with wider variations, of a glissando or melodies.

The combination of these two variations constitutes fixed-waveform synthesis with time-varying amplitude and frequency. The waveform is fixed, while the amplitude and frequency vary. The partials are exact multiples of the fundamental, and they all behave the same.

Fixed-waveform synthesis is realized rather simply. Hence, it is often employed when good sound quality is not required. The constant waveform gives the sound a mechanical, dull, and unnatural character, which soon annoys the audience. Thus,

in musical applications, fixed-waveform synthesis is not very effective when used alone. It is employed for its simplicity when timbral variety is not required, for example, for real-time synthesis on very limited hardware.

For economy, other methods of generating waveforms that do not use tables or multiplications have been devised. The simplest generates a square or (more generally) a rectangular wave, alternating sequences of positive and negative samples of the same value. The frequencies that can be obtained are submultiples of the sampling rate.

A sawtooth signal can also be generated by an ac-

cumulator to which a constant value is continuously added. The output increases linearly until it overflows and starts from the beginning. The signal frequency is proportional to the constant value. This method is used to produce linearly variable control signals. Every time the additive constant changes, the slope changes. Hence, functions composed of straight segments, such as envelopes, can be obtained.

This technique has been generalized recently by Mitsuhashi (1982a). A polynomial of degree N can be generated by putting N accumulators in cascade. The accumulators are initialized by the value of the forward differences, in decreasing order, of the polynomial to be generated (Cerruti and Rodeghiero 1983). The waveforms obtained exhibit great variety and, in certain conditions, they are periodic.

Granular Synthesis

The technique of fixed-waveform synthesis produces rather static sounds in time. Yet a fundamental characteristic of musical sound is its timbral evolution in time. A sound can be thought of as a sequence of elementary sounds of constant duration, analogous to a film, in which a moving image is produced by a sequence of images.

In computer music, the elementary sounds are called grains, and the technique of exploiting this facility is granular synthesis (Roads 1978). The grains can be produced by a simple oscillator or by other methods. The duration of each grain is very short, on the order of 5-20 msec.

There are two ways to implement granular synthesis. The first is to organize the grains into frames, like the frames of a film. At each frame, the parameters of all the grains are updated. This is the approach sketched by Xenakis (1971). The second way involves scattering the grains within a mask, which bounds a particular frequency/amplitude/time region. The density of the grains may vary within the mask. This is the method imple-

mented by Roads (1978).

A problem with granular synthesis is the large amount of parameter data to be specified. In some other types of synthesis (additive and subtractive, to be discussed shortly), these data can be obtained by analyzing natural sounds. However, no analysis system for granular synthesis has been developed. Another possibility is to obtain the parameter data from an interactive composition system, which allows the composer to work with high-level musical concepts while automatically generating the thousands of grain parameters needed.

Additive Synthesis

In additive synthesis, complex sounds are produced by the superimposition of elementary sounds. In certain conditions, the constituent sounds fuse together and the result is perceived as a unique sound. This procedure is used in some traditional instruments, too. In an organ, the pipes generally produce relatively simple sounds; to obtain a richer spectrum in some registers, notes are created by using more pipes sounding at different pitches at the same time. The piano uses a different procedure. Many notes are obtained by the simultaneous percussion of two or three strings, each oscillating at a slightly different frequency. This improves the sound intensity and enriches it with beatings.

In order to choose the elementary sounds of additive synthesis, we first note that the Fourier analysis model enables us to analyze sounds in a way similar to the human ear and so to extract parameters that are perceptually significant. When we analyze a real, almost-periodic sound, we immediately notice that each partial amplitude is not proportionally constant, but that it varies in time

Fig. 3. Additive synthesis.

AI(t) f,(t) A2(t) f2(t) AM(t) fM(t)

$s(t)$

according to different laws. In the attack portion of a note, some partials, which in the steady state are negligible, are often significant.

Any almost-periodic sound can be approximated as a sum of sinusoids. Each sinusoid's frequency is nearly multiple that of the fundamental, and each sinusoid evolves in time. For higher precision, the frequency of each component can be considered as slowly varying. Thus, additive synthesis consists of the addition of some sinusoidal oscillators, whose amplitude, and at times frequency, is time varying (Fig. 3).

The additive-synthesis technique also provides good reproduction of nonperiodic sounds, presenting in the spectrum the energy concentrated in some spectral lines. For example, Risset (1969) imitated a bell sound by summing sinusoidal components of harmonically unrelated frequencies, some of which were beating. In Risset's example, the exponential envelope was longer for the lower partials.

Additive synthesis provides great generality. But a problem arises because of the large amount of data to be specified for each note. Two control functions for each component have to be specified, and normally they are different for each sound, depending on its duration, intensity, and frequency. The possibility of data reduction has been investigated. At Stanford University, a first result has been obtained by representing the control functions of the amplitude and the frequency of each component by line segments, without affecting "naturalness" of the sound (Grey and Moorer 1977).

The next step has been to investigate the relations between these functions (Risset and Mathews 1969; Beauchamp 1975) or their relation to others of more general character (Charbonneau 1981). Additive synthesis is most practically used either in synthesis based on analysis (analysis/synthesis), often transforming the extracted parameters, or

when a sound of a precise and well-determined characteristic is required, as in psychoacoustic experiments. In any case, in order to familiarize musicians with sound characteristics and frequency representations, the technique is also useful from a pedagogical point of view.

Additive synthesis can be generalized by using waveform components of other shapes besides sinusoids. To allow the reproduction of any sound, these waveforms have to satisfy specific mathematical properties. Walsh functions are an example of this kind of function; they are used for their simple hardware realization (Rozenberg 1979).

VOSIM

In the synthesis techniques already discussed, oscillators that periodically reproduce a given waveform are employed. Other synthesis techniques, instead of continuously repeating a given waveform, calculate it anew each period, with minor variations. The control of this calculation process allows continuous spectral variations. A common method of this type is the voice simulation (VOSIM) technique. A VOSIM oscillator has been devised in a project at the Institute of Sonology in Utrecht (Kaegi 1973, 1974; Kaegi and Tempelaars 1978).

The VOSIM waveform (Fig. 4) consists of a sequence of N pulses of shape sin2, of the same duration T, and of decreasing amplitude. The sequence is followed by a pause M. Each pulse's amplitude is smaller than the preceding one, by a constant factor b.

The VOSIM spectrum (Fig. 5a) is described as the product of two terms (Tempelaars 1976; De Poli and De Poli 1979). The first term S, (Fig. 5b) depends only on the pulse shape and limits the signal bandwidth to 2F (being F = 1/T). The second term S2 (Fig. 5c) depends on the relationship between the individual pulse amplitudes. S2 is periodic in the frequency domain with a period F, and it is sym-

Fig. 4. VOSIM oscillator: T is the duration of single pulse, M the rest between two sequences of pulses.

Fig. 5. Spectral envelope of a VOSIM oscillator (N = 5, b = 0, 8) (a). The envelope is the product of the terms S, (b) and S2 (c).

1.5
1
0.5
0 5 M- 10 15
T

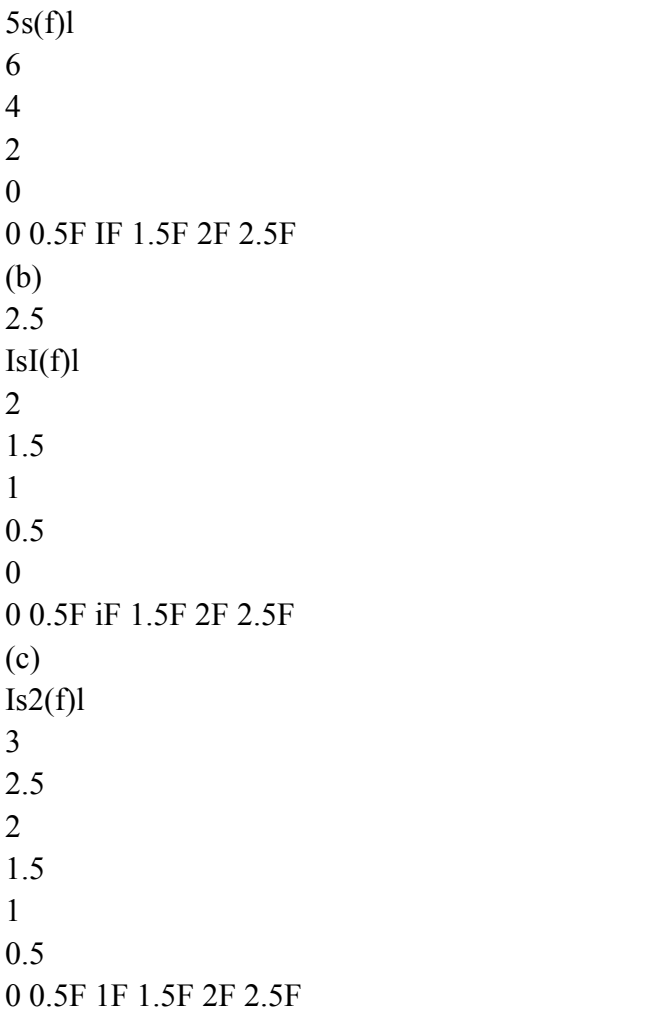
metric with respect to F/2. When b 1, its amplitude will be greater around the extremes of the period 0 and F. When b - -1, its amplitude will be greater in the central position around F/2. Thus, a characteristic formant in F or F/2 will result. The number of pulses N produces N oscillations in the S2 term between 0 and F, with strong signals for b near +F.

This constitutes the spectral envelope of the repeated waveform. Taking a as the ratio between the signal period and a single pulse duration, the number of the harmonic corresponding to the formant is a if b is positive, and a/2 if b is negative. Thus, by varying a, the formant shifts, and the relative amplitude of all the harmonics vary continuously but not homogeneously, following the spectral envelope. The signal and the formant frequencies can be separately controlled.

More kinds of sounds can be obtained by modulating (sinusoidally or randomly) the value of the time interval M between two consecutive pulse sequences. This means that a varies independently from T. In this case, the formant frequency remains constant while the harmonic amplitudes vary. Then

the ear can easily perceive the spectral envelope and fuse the components together. This property makes the VOSIM oscillator effective in musical applications.

If a variation is strong, practically aperiodic sounds or colored noises are obtained. Adding several VOSIM oscillators allows one to control the position of the formants. This results in an additive



synthesis of already complex sounds rather than of sinuosidal components. Instead of the frequency of partials, the position of the formants is controlled. This is a more relevant parameter, from an acoustic

standpoint.

The formant-wave-function synthesis of Rodet (1980) is analogous to VOSIM, but it allows overlapping of single waveforms. This provides better control and generally richer sounds. Mitsuhashi (1982a) and Bass and Goeddel (1981) generalized the VOSIM model by including the case of pulses of any amplitude and using different elementary waveforms.

Synthesis by Random Signals

Up to now, we have considered signals whose behavior at any instant is supposed to be perfectly knowable. These signals are called deterministic signals. Besides these signals, random signals, of unknown or only partly known behavior, may be considered. For random signals, only some general characteristics, called statistical properties, are known or are of interest. The statistical properties are characteristic of an entire signal class rather than of a single signal. A set of random signals is represented by a random process. Particular numerical procedures simulate random processes, producing sequences of random (or more precisely, pseudorandom) numbers. The linear congruential method is commonly used to produce uniformly distributed numbers. From a starting value X0, a sequence of random integers X0, X1, . . . , XK... is generated according to the relation

XK+ = (a XK + C)modm,

where m is the modulus and the maximum sequence period, and a and c are two specific integer constants.

The modulus operation can be avoided by choosing m as the maximum number representable in the computer, that is, m = 2b, where b is the word length (bit number in a binary computer). So the numbers are automatically truncated. The choice of X0, a, and c greatly affects the statistical characteristics of the generated sequence, and its acceptability has to be accurately verified by statisti

tests. A general discussion of various distribut
and the methods used to generate them can be
in Lorrain's paper (1980).
Random sequences can be used both as sign
(i.e., to produce white or colored noise used as
put to a filter) and as control functions to pro
a variety in the synthesis parameters most per
tible by the listener.
In the analysis of natural sounds, some chara
teristics vary in an unpredictable way; their m
statistical properties are perceptibly more sign
cant than their exact behavior. Hence, the add
of a random component to the deterministic f
tions controlling the synthesis parameters is o
desirable.
In general, a combination of random processe
is used because the temporal organization of th
musical parameters often has a hierarchical asp
It cannot be well described by a single random
process, but rather by a combination of rando
cesses evolving at different rates.

Linear Transformations

Let us now examine techniques for signal modifica-
tion. A transformation is a set of rules and proce-
dures transforming a signal called input to another
signal called output. A transformation is linear if
the superimposition principle is valid, that is, if the
effect of the transformation caused by a two-signal
addition is equal to the addition of the individual
signal transformations applied separately. In partic-
ular, in a linear transformation a signal can be mul-
tiplied by a constant but not by another signal.
Digital filters are linear transformations that can
be described by the following difference equation:

N M
Sa, yK=0 i1=0
where aK and b-
and y(i) are the i
signal. The value
ear combination

with the precedi
DePoli 13

Fig. 6. Finite-impulse-response (FIR) filter with two zeros described by the equation y(n) = x(n) + a,x(n - 1) + a2x(n - 2)
(a). Infinite-impulse-response (IIR) filter with two poles described by the equation y(n) = x(n) + P1y(n - 1) + 32y(n - 2)
(b).
x(n) (a)y(n)
Z-1
x(n - 2)
----'--
x(n) (b)y(n)
Z-1
- ~ y(n- 2)

the input is sinusoidal, the steady-state output is sinusoidal with the same frequency. The amplitude and phase of the frequencies are determined by the system. That is why this transformation is called a filter.

Subtractive Synthesis

Sound produced by filtering a complex waveform is called, sometimes inappropriately, subtractive synthesis. First, a periodic or aleatoric signal rich in harmonics is generated by the previously examined techniques or others. This signal must contain energy in all frequencies required in the output sound. Second, one or more filters are used to alter selectively the specific frequency components. The undesired components are attenuated (subtracted) and others are eventually amplified. When the filter coefficients change, the frequency response changes, too. Thus, it is possible to vary characteristics of

the output sound.

In modular diagrams, filters are usually represented by rectangles and the difference equation or the transfer function is given as a label near the rectangle. Two examples of simple digital filters, showing their internal structure, are shown in Fig. 6. The first filter (Fig. 6a) has a finite-impulse response (FIR). This structure is useful to produce transmission zeros: that is, it can nullify some frequencies that depend on a1, a2 values and on the sampling rate. The second filter (Fig. 6b) is recursive, or has an infinite-impulse response (IIR). Feedback in the structure amplifies certain frequencies, that is, produces transmission poles. When used as bandpass filter, in general terms, the coefficient P, controls the center frequency and the coefficient 32 the bandwidth.

One of the most attractive aspects of digital filtering is that it is analogous to the functioning of many acoustic musical instruments. Indeed, instrument physics can be used as a model for synthesis. For example, in the brasses and woodwind instruments, the lips or vibrating reed generate a periodic signal rich in harmonics. The various cavities and the shape of the instrument act as resonators, enhancing some spectral components and attenuating others. In the human voice, the excitation signals are periodic pulses of the glottis (in the case of voiced sounds) or white noise (in the case of unvoiced sounds-for example, the consonants s and z). The throat, the mouth, and the nose are the filtering cavities, and their dimensions vary in time. Their great variability makes the human voice the most rich and interesting musical instrument.

Today, subtractive synthesis is the standard means of speech synthesis. An analysis procedure, called linear predictive coding (LPC), allows us to obtain

Fig. 7. Elementary filters
used in reverberators.
Comb filters (a). All-pass
filter (b).

the pitch and the coefficients of a recursive (poles only) filter (see Cann's [1979-1980] tutorial and Moorer's paper [1979a]). These data can be utilized to synthesize the sound directly or following modification. For example, speech can be accelerated or slowed down, and pitch can be varied. An instrument or orchestral sound can be used as input to the filter, producing the effect of a "talking orchestra." Interesting possibilities for musique concrete sound processing arise. Not only simple filtering of sounds is possible, but the modification of their most intrinsic characteristics is also made possible by varying the parameters of the deduced sound-production model.

Generally, LPC is relatively difficult to use. Intuitively, the filter characteristics depend on the position of the zeros and the poles in the transfer function. These characteristics are affected in a complex and nonintuitive way by the filter coefficients. In some simple cases, approximate formulas give the coefficients as functions of significant parameters, that is, center frequency and bandwidth, or cutoff frequency and slope. The filters can be used in series or in parallel. In the most complex cases, a precise analysis is obtained by using specific programs for digital filter design and analysis. Such digital filters can be very stable and precise, but only at the cost of a large amount of calculation. Simple linear digital networks can also be used as oscillators (Tempelaars 1982) by applying a pulse sequence to the input and choosing an impulse response equal to the signal function to be generated.

Reverberation

One application of digital filters is sound reverberation. An acoustic environment can be simulated by

distributing sound among different loudspeakers and by adjusting the ratio between direct and reverberated sound (Chowning 1971). Most of the studio reverberators sold today use digital technology.

The two elementary filters used in reverberation are shown in Fig. 7. The first filter is called a comb filter; in it, the signal is delayed a certain number of samples, attenuated, and added to the input. An exponentially decaying, repeated echo is so obtained. The frequency response is characterized by equispaced peaks-hence this filter's name. The peaks' amplitude increases as G approaches 1.

(a)
+)(Delay
(b)
- G

The second filter is called an all-pass filter, since the frequency response is flat and there is only a phase shift. The input signal is attenuated and subtracted from the delayed signal so that the feedback effect is compensated and the echoes are maintained. The all-pass property is valid only in the steady state with stationary sounds, not in transient states. Thus, it has a well-defined sound quality that a skilled listener can easily distinguish.

Reverberators are built combining some of these filters (Moorer 1979b). Distinguishable signal repetitions should not occur in them, since the reverberated result should consist of a diffused sound. The delay time of each elementary filter has to be chosen very carefully. Sometimes a nonrecursive echo generator is added to produce the first aperiodic echoes, which are the main perceptual determinants of the characteristics of the room.

Nonlinear Techniques

In addition to linear transformations, which are used in other fields and have a rather developed theory, nonlinear transformations are used more and

Fig. 8. Waveshaping.

more commonly in musical applications. They derive mainly from electrical communication theory, and they have proved to be promising and effective. One use of nonlinear synthesis is in the large amount of computer music generated by frequency modulation (FM) synthesis (Chowning 1973).

In the classic case, nonlinear techniques use simple sinusoids as input signals. The output is composed of many sinusoids, whose frequency and amplitude depend mostly on the input ones.

Two main types of nonlinear techniques can be distinguished, waveshaping and modulation. In waveshaping, one input is shaped by a function depending only on the input value in that instant. In modulation (with two or more inputs), a simple parameter of one signal, called the carrier, is varied according to the behavior of another signal, called the modulator. In electrical communications (e.g., radio) the spectra of the signals are clearly distinguished and therefore easily separable. The originality in computer music application is the utilization of signals in the same frequency range. Thus, the two signals interact in a complex way, and simple input variation affects all the resultant components.

Often, the input amplitudes are varied by multiplying them by a constant or time-dependent parameter I, called the modulation index. Thus, acting only on one parameter, the sound characteristics are substantially varied. Dynamic and variable spectra are easily obtainable. In additive synthesis, similar variations require a much larger amount of data.

Waveshaping

A linear filter can change the amplitude and phase of a sinusoid, but not its waveform, whereas the aim of waveshaping is to change the waveform. The distortion of a signal heard from a nonlinear ampli-

fier is common. The output from a nonlinear amplifier of a sinusoidal signal is a signal with the same period, but with a different waveform. The various harmonics are present, and their amplitude depends on the input and on the distortion. In stereo systems, these distortions are usually avoided,

x(t)

y(t)y~t)

while waveshaping (Arfib 1979; Le Brun 1979; Roads 1979) exploits them to generate periodic sounds, rich in harmonics, from a simple sinusoid. The function F(x), describing distortion, is called the shaping function, and it associates with each input value the corresponding output value independent of time. If the input is x(t) = cos(27r ft), the output is

s(t) = F(x(t)) = F(cos[2rr ft]).

In analog synthesis, it is difficult to have an amplifier with a precise and variable distortion characteristic. In digital synthesis, this technique is extremely easy to implement (Fig. 8). As in the case of the oscillator, the shaping function can be previously computed and stored in a table. All that is necessary is to look up the proper value from the table.

Generally, if F(x) = F,(x) + F2(x), the distortion produced by F is equal to the sum of those produced by F, and F2 separately. Usually, the shaping produces infinite harmonics. But when a polynomial of degree N is chosen as shaping function, only the first N harmonics are present. Thus, foldover is easily avoided. Arfib and Le Brun deal extensively with the mathematical relations among the coefficients di of the shaping polynomial and the amplitudes hi of harmonics generated when the amplitude I of the cosinusoidal input varies.

The shaping function, producing the jth harmonic, is the Chebychev polynomial 7T(x) of degree i (Fig. 9). Thus, to obtain the various harmonics of

Fig. 9. Chebychev poly-
nomial of degree K used as
shaping function produces
only the Kth harmonic. In
the figure, K = 3.
T3(cos[ot]) = cos(3ot)
T3(x) = 4X3 - 3x
X = cos(wt) wt
01 1
-rr/6 -
2I/37T/3_ os(ut)
73/2
27r/3
1lT/6
77r/6
137T/6
ot
amplitude h,, it is sufficient to add the correspon-
dent Chebychev polynomials, each multiplied
by hi:
N N
F(x) = h(x) = d;x.=0 i=O0
From these relations, it follow
monics are comp
even polynomial
harmonics. In th
only the odd ha
cient of x7 affect
enth harmonics
harmonic of ord
odd) coefficients
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example, the seventh harmonic is affected by the
odd coefficients from the seventh up to the degree
of the polynomial.
When the input amplitude I varies, the distortion

and the output spectrum vary. This is similar to an expansion or contraction of the function, since greater or smaller range of the function is employed. From a mathematical point of view, the amplitude variation corresponds to the multiplication of each polynomial coefficient d, by II. The amplitudes of the even or odd harmonics depend on I according to the even (or odd) polynomials, which contain the terms from the harmonic order up to the polynomial degree.

If the spectrum is rather smooth, the number of significant harmonics increases with the index. Thus, a typical characteristic of real instruments is reproduced, in that amplitude and spectrum are correlated. The amplitude and loudness of the output vary with the input amplitude. In simple cases, this effect can be compensated for by multiplying the output by a suitable normalization function. But in musical applications, the amplitude of the signal is rarely constant, and it is multiplied by an envelope. Normalization can be avoided by combining it with the amplitude envelope in experimental or intuitive ways after considering the normalization function.

It is also advisable to choose the even (or odd) polynomial coefficients with alternating signs, that is, according to the following model: + + - - + + - -. It is also advisable that the hi amplitude not decrease abruptly, sharply limiting the band. Otherwise, a spectrum would result that varied very irregularly with I.

Dynamic spectral behavior cannot be easily anticipated from the coefficients or from the static spectrum. Moreover, the same (absolute-value) spectrum can be produced by many polynomials with different dynamic behaviors (Forin 1982). With waveshaping, listening and graphic considerations have more relevance than purely mathematical formulations.

Another dynamic variation of waveshaping that is easy to implement occurs when a constant is

added to the input; the shaping function shifts horizontally. Even in this case, the spectrum varies. The signal is periodic, with the same number of harmonics. But in this case, the harmonic behavior depends on both the even and the odd coefficients. Generalizations of waveshaping technique are possible. Reinhard (1981) studied the relations that produce the partials generated by the polynomial distortion of two cosine waves of frequency f, and f2. All the components of frequency 1K f, ? Jf2 with IK + j|l N, where N is the polynomial degree, are present.

Shaping functions that are not polynomial can be used if the spectra produced by them are almost band limited. Of particular interest is the use of trigonometric and exponential functions (Moorer 1977) and of those where the input also appears in the denominator (Winham and Steiglitz 1970; Moorer 1976; Lehmann and Brown 1976; De Poli 1981).

Due to the wide spectral variation induced by only one parameter (amplitude or shift), waveshaping is particularly convenient in musical applications, especially in combination with multiplicative synthesis. Moreover, it is suitable for modeling the sound production of some acoustic instruments (Beauchamp 1979, 1982). There is a large and not intuitive problem in choosing the coefficients, however, and further research is required.

Multiplicative Synthesis (Ring Modulation)

The simplest nonlinear transformation consists of the multiplication of two signals. In analog synthesizers, it is called ring modulation (RM). Sometimes it is also called amplitude modulation (AM), but the two differ, especially in their realization. With two inputs x,(t) and x2(t), the output is s(t) = Xl(t) . x2(t). Obviously, when the inputs interchange, the result does not vary. The resulting spectrum is obtained from the convolution of the two signals' spectra. Usually, one of the two signals,

called the carrier, is sinusoidal; the result is not too complex and noisy.

When x1 is the sinusoidal carrier of frequency f/, and x2 (modulator) is sinusoidal with frequency f2, from cos(a) , cos(3) = 12lcos(a + 3) + cos(a - /)1,

Fig. 10. Multiplicative synthesis. Spectrum of a periodic signal X2 with four harmonics (a). Resulting spectrum when d2 is multiplied by a sinusoid of frequency f, greater than its bandwidth (f, = 7f2) (b). Resulting spectrum when x2 is multiplied by a sinusoid of frequency inferior to its bandwidth (f, = 26f2) (c). The components deriving from the folding of negative frequencies are shown as dashed lines.

the output consists of two sinusoidal partials of frequency fI + f2 and f, - f2. The phases of the output are also the sum and the difference of the phases of the two inputs. For example, if x, and x2 frequencies are 400 Hz and 100 Hz, the output has two partials of frequency 500 Hz and 300 Hz.

Negative frequencies may occur, for example, when f, = 100 Hz and f, = 400 Hz. This often happens in modulations (foldunder) and can be explained by the trigonometric relation cos(a) = cos(-a), from which cos(27r ft + (4) = cos(7r[-flt - 4). The alteration of the frequency sign only changes the sign of the phase with respect to the cosine. In particular, a cosine signal is unaffected, while a sine wave changes its sign. In the interpretation of the results, only absolute frequency

values have to be considered. Usually, the phase is not significant, as the ear is not terribly sensitive to it. But the phase has to be taken into account while summing the amplitude of components of identical frequencies.

In multiplicative synthesis, usually x2 is periodic with frequency f2. The multiplication causes every harmonic spectral line of frequency K . f2 in the original signal to be replaced by two spectral lines (called sidebands) of frequency f, + K f2 and f, - K f2. The resulting spectrum has components of frequency If +? K f2 , where K is equal to the order of the different harmonics in x2 (Fig. 10). Thus two sidebands, symmetric with respect to the carrier, occur. When f, is less than the greatest frequency in x2, then the negative frequencies fold around zero, as discussed above.

The possibility of shifting the spectrum is very intriguing in musical applications. From simple components, harmonic and inharmonic sounds can be created, and various harmonic relations among the partials can be established. If x2 is a signal with spectrum X2, the signal obtained from its multiplication with a sinusoid of frequency f, has two sidebands symmetric with respect to f, and shaped like X2.

A periodic signal x, can be expanded in Fourier series. Each x, partial will have sidebands of amplitude proportional to its own. If f1 is less than the bandwidth of x,, then the sidebands overlap with

(a) jx2(f)l

(b)

I|S(f)l

ff2

(c)

S(f)I

Iu I li I fl

eventual component superimposition. In this case, the phases have to be taken in account while summing. Dashow (1978, 1980) describes some general-

ization of this technique and employs the generated spectra for particular "harmonizations" of pitches specified by the composer.

Amplitude Modulation

In RM, the carrier does not appear in the spectrum created by the product of a sinusoidal carrier with another signal, except when the modulator has a direct current (dc) component. In carrying out the modulation in AM (Fig. 11), the carrier is present in the output, with an amplitude independent of the sidebands. The formula for AM is as follows:

s(t) = xl(t) . (K + x2(t)).

The result is RM with carrier added. When the carrier is sinusoidal and the modulator is periodic, the spectrum is composed of partials of frequency If1 1 K f,2, with K = 0, 1, . It is useful to distin-

Fig. 11. Amplitude modulation.

K A2 f2 f1

X2(t)

x1(t)

s(t)

guish between the two modulations because they have different realization schemes.

Spectra of Type If , K f2

The following considerations are valid for all spectra whose components are of type If, + K f2h, with K = 0, 1, ... The spectrum is characterized by the ratio f,/f2. (This is often referred to as the carrier-to-modulator [c:m] ratio.) When this ratio is rational, it can be expressed as an irreducible fraction fI/f2 = NI/N2, with N, and N2 as integers that are prime between themselves. In this case, the resulting sound is harmonic, since the various components are a multiple of a fundamental according to integer factors. The fundamental frequency is

fo = - /

N1 N2 '
and the carrier coincides with the NIth harmonic. If N2 = 1, all the harmonics are present and the sideband components coincide. If N2 = 2, only odd harmonics are present and the sidebands superimpose. If N2 = 3, the harmonics that are multiples of 3 are missing. The c: m ratio is also an index of the harmonicity of the spectrum. The sound is mo "harmonious" intuitively when the N /N2 rati simple and formally when the N, N2 productsmaller. The ratios can be grouped in families (Truax All ratios of the type If, K f2l/f2 can produce same components that flf, produces. Only th tial coinciding with the carrier (f,) changes. Fo ample, the ratios 2/3, 5/3, 1/3, 4/3, 7/3 and s all belong to the same family. Only the harmo that are multiples of 3 are missing (see N2 = 3 the carrier is respectively the second, fifth, fir fourth, seventh, and so on harmonic.

The ratio that distinguishes a family is defin normal form when it is - 1/2. In the previous ample, it is 1/3. Each family is characterized b ratio in normal form. Similar spectra can be p duced using ratios from the same family. Diffe spectra are obtained by sounds of different fa When the fl/f2 ratio is irrational, the resulti sound is aperiodic and hence, inharmonic. Of p ticular interest is the case of an f,/f2 ratio app mating a simple value, that is,

fl/f2 = N1/N2 + e.

Here the sound is no longer rigorously periodi The fundamental frequency fo is still f2/N2, the harmonics are shifted from their exact valu by +e / f2. When N2 is equal to 1 or 2, the po and negative components are not superimposed beat with a frequency of 2e / f2. Hence, a smal of the carrier does not change the pitch, even slightly spreads the partials and makes the sou more lively. But the same shift of the modula frequency f, changes the sound's pitch.

Frequency and Phase Modulation

Another type of modulation, suggested by Cho
ing (1973), has become one of the most widely
synthesis techniques. In general, it consists of
modulation and it can be realized both as ph
modulation (4M) or as FM. This technique do
not derive from models of production of phys
sounds, but only from the mathematical prope
ties of a formula. It has some of the advantage

Fig. 12. The number of sig-
nificant sidebands in FM.

waveshaping and RM, and it avoids some of their
drawbacks.

The technique consists of the modulation of the
instantaneous phase or frequency of a sinusoidal
carrier according to the behavior of another signal
(modulator), which is usually sinusoidal. It can be
expressed as follows:

s(t) = sin(2zT f t + I sin[27r fmt]) =

=>K JK (I)sinj27T(f, + Kfj)tI.

The resulting spectrum is of the type If, +? K f,,. All
the spectral considerations discussed previously are
applicable, particularly those regarding negative fre-
quency, foldunder, fclf, ratios, and harmonic andinharmonic sounds.

The amplitude of each Kth side component of
the FM technique is given by the Bessel function of
Kth order computed in I. To plot the spectrum, a
table of Bessel functions has to be referenced to ob-
tain the amplitudes of the carrier and of the side
frequencies in the upper sideband. The odd-order
side frequencies in the lower sideband have signs
opposite to those in the upper one, and the even-
order side frequencies have the same sign. The
negative frequencies, being sine waves, are folded,
changing the sign. When superimposition occurs,
the amplitudes are added algebraically.

When I (called the modulation index) varies, the

amplitude of each component varies as well. Thus, dynamic spectra can be obtained simply by varying this index. Each component varies its amplitude by following the corresponding Bessel function. A Bessel function can be asymptotically approximated by a damped sinuosoid. So when the index varies, some components increase and others decrease, all without sharp variations.

In Eq. (1), the sum includes infinite terms, so theoretically the signal bandwidth is not limited. But, practically, it is limited. In the Bessel function's behavior, only a few low-order functions are significant for small index values. When the index increases, the number and the order of the significant functions increase. For a given index, the side amplitudes oscillate with gradually increasing amplitude and slowly increasing period all the way from

25
20 /
15 /
5
M ' I I
I 5 10 15 20

the origin to a
toward zero. Th
slightly below
Usually, in the
signal, all side f
than /loo of th
ered. The numb

M = I + 2.4 J10.27

(See Fig. 12.) Often, as a rule of thumb, it is roughly considered as

M = I+ 1.

In Eq. (1), the sum can be performed for K fr
-M to +M. For a harmonic sound, that is, w
the ratio fc/fm = NI/N2 is simple, the maxim
der of significant harmonics is N, + M ' N2.
For wide index variations, the sounds produc
are characteristic of the FM technique. A typi

timbre of FM sound is easily recognizable an
well defined. This does not happen for small i
variations or for compound carriers or modul
Frequency modulation synthesis has another p
DePoli 21

Fig. 13. Frequency
modulation.
A(t) fc d(t) fm
s(t)
Fig. 14. Frequency modu-
lation with N carriers
modulated by the same
oscillator.
a, f/ a2 f2 aN fN d f,
+ + +
s(t)
erty that is very important in musical applications:
the maximum amplitude and the signal power do
not vary with the index I. Unlike the situation in
waveshaping, normalization of the output is not
necessary.
Let us now examine the difference between OM
and FM. Phase modulation is defined as follows:
s(t) = sin (2M ft + 0[t]),
and it corresponds to Eq. (1) if the modulating sig-
nal is 0(t) = I sin(2rr f, t).
Frequency modulation occurs when the instanta-
neous frequency varies around the carrier value ac-
cording to the behavior of the modulating wave. For
a signal s(t) = sin(p[t]), the instantaneous frequency
is fj = (1/27r) (di[t]l/dt). Thus, the instantaneous
frequency of the signal in Eq. (1) is as follows:
fi = fc + I fm , cos(2r f,, t).
The frequency varies around f, with a maximum
deviation d = I f,. Thus, with a modulating wave
I - f, cos(27r f, t), an FM equivalent to OM is ob-
tained. Both phase and frequency modulations are
special cases of angle modulation.

In sound synthesis programs, frequency-driven oscillators are provided. The integration involved in calculating the instantaneous phase is therefore computed automatically. Frequency modulation is normally implemented as in Fig. 13. A change of the phase between the carrier and the modulating wave in Eq. (1) only changes the reciprocal phase of the partials. If components superimpose, their total amplitude changes, and a direct-current component may appear. The next sections examine some useful extensions of the basic algorithm.

Nonsinusoidal Carrier

Here we consider a periodic nonsinusoidal carrier. The result of its modulation is the modulation of each of its harmonics by the same wave. Sidebands of amplitude proportional to each harmonic will be present around the carrier. The result is a spectrum with components of frequency In f, + K . fmi,, with K = 0,... , M and n = 1,..., N, when Nis the number of significant harmonics. The maximum frequency present is N - f, + M . f,. In general, there may be various independent carriers modulated by the same wave (Fig. 14) or by different modulating signals. This is like additive synthesis, only instead of sinusoidal addends, more complex addends are used. For example, harmonic sounds can be generated by controlling the various spectral ranges with a few significant and independent parameters. Sounds of the same "family" are possible.

The frequency of each carrier determines the

Fig. 15. Frequency modulation with two modulators.

Fig. 16. Frequency modulation with N modulators.

A fc dl(t) f, d2(t) f2

s(t)

location of the formant position, the amplitude determines its energy, and the modulation index specifies its bandwidth. Chowning (1981) demonstrated these facilities in synthesis of the singing voice of a soprano.

Compound Modulation

Let us examine the case of a modulation composed of two sinusoids (Fig. 15), each with its own modulation index, applied to a sinusoidal carrier. The formula for two-sine-wave OM (Le Brun 1977) is as follows:

s(t) = sin(121T fct + I, sin[27r f t] + 12 sin[27r f2t])
= K n, JK(I1) Jn(12) . sin(I27T[fc + K f, + n f2ltI).

The same result can be obtained with FM using as modulating signal the following expression:

I,f,/cos(27r f, t) + I2f2cos(2r f2t).

The resulting spectrum is much more complex than in the one-modulator case. All the components of frequency If/ ? K f, n f,2 are present, and their amplitude is JK(I) " Jn(I2).

To interpret the effect, let us consider f1 > f2. If only f, were present, the resulting spectrum would have a certain number of components of amplitude JK(I1) and frequency fc + K f1. When the modulator

A fc d1(t) f, d2(t) f2 dM(t) fM
s(t)

f, is applied, these components become carriers, with sidebands produced by f,. The resulting bandwidth is approximately equal to the sum of the two bandwidths.

If the frequencies have simple ratios, the spectrum is of the type If, -K f,,, where now f, is the greatest common divisor of f, and f,. For example, with f, = 700 Hz, f = 300 Hz, and f2 = 200 Hz, the components are 1700 ? K 1001. Thus, by choosing f, and f2 multiples of f,, sounds belonging to the same family as a simple modulation, but with a more complex spectral structure, can be generated.

In general, if the modulating signal is composed of N sinusoids (Fig. 16), the following relations hold:

s(t) = sin (2rT fct + EI sin[2r fs t]
N
= KS, HKS (Is) sin|2ir(fct + Y Ifs)t1.
Thus, all the components of frequency If , Klf/ -. . ? KNfNI, with amplitudes given by the product
of N Bessel functions, are obtained. A very complex spectrum results. If the relations among the frequencies f, are simple, that is, if the modulating wave is periodic, then the spectrum is of the type Ifc ? K fm, where f,m is the greatest common divisor among the modulating components. Otherwise, the sonorities are definitely inharmonic and particularly noisy for high indexes.

Fig. 17. Nested FM.

Nested or Complex Modulation

Let us examine the case of a sinusoidal modulator that is phase modulated by another sinusoid. The signal is defined as follows:

s(t) = sin(I27r ft + I sin[2ir f, t + 12 sin{2ir f2t}]))
= K JK(Il)sin(I2zr[fc + K f,]t
+ K 12 sin[27r f2t]I)
= 1K J(I,)" Jn(KI2)sin(2r[fc + K,f, + n f2]t).

The result can be interpreted as if each partial produced by the modulator f, were modulated in its turn by f2 with modulation index K I,. Thus, all the partials of frequency If +? K f, n f2j, with approximately 0 - K - I, Os n In , 1 I2, are present. The maximum frequency is fc + I1I(f + 12/2).

The structure of the spectrum is similar to that produced by the two-sinusoid modulation, but with a larger bandwidth. Even where fm is the greatest common divisor between f, and f2, the spectrum is of the type If, + K fn-.

In the equivalent realization by FM (Fig. 17), the spectrum is of the same type, but with slightly different amplitudes. A direct-current component in the resulting modulating wave added to the carrier

is avoided by choosing a sine wave modulated by a cosine wave.

This technique is made more interesting by an algorithm suggested by Justice (1979), which enables an analysis of a sound according to this model, with the frequency and the index behavior of two or more nested modulators being deducible.

Other Two-Input, Nonlinear Transformations

Mitsuhashi (1980) proposed a more complex two-input, nonlinear transformation, in which the instantaneous phase and amplitude of an approximately sinusoidal signal are simultaneously varied. In another paper, Mitsuhashi (1982c) generalized this technique while discussing some criteria in choosing the two-input, nonlinear function and suggesting two examples. The function is time independent, bidimensional, and considered periodic outside the definition field. Thus, it can be implemented with a two-dimensional table, with analogy to an oscillator. This technique appears very inter-

A fc d (t) f/ d2(t) f,

+

s(t)

esting, even if it seems to be difficult to find a simple expression that bounds significant parameters of the resulting spectrum to the input and function characteristics. Another promising modulation technique is linear sweep synthesis, recently suggested by Rozenberg (1982).

Conclusion

As a consequence of progress in digital hardware and software, the initial antithesis between computing efficiency and timbral richness is lessening. Digital sound quality largely depends on the amount of introduced or controlled detail; excessive simplifications lead often to trivial results. It follows that increased computing power can generate more sophisticated results.

A musically interesting sound can be obtained in two ways. The first consists of the utilization of

more complex techniques or of the combination of

many of the techniques described here. Many linear and nonlinear transformations are possible. Most of the parameters do not have to be constant and can be varied by control functions and random signals.

The other synthesis approach consists of the superimposition of many simple sounds produced by basic techniques. The evolution of the individual sounds is not complex, and the richness of the result essentially depends on their combination. In this approach, the parameters of many elementary sounds have to be given. Specific programs are often used to define these parameters.

Sound evolution can be regulated either by control functions in the synthesis or by programs computing the parameters for the synthesis. In any case, many details of the sound have to be accurately controlled. Their coherence both within the sound and in the context of adjacent and simultaneous notes has to be guaranteed. The relations among sounds can be more easily highlighted when they are reflected not only in macroscopic parameter variations but also in internal structure.

The extensive utilization of a single technique reveals its peculiar characteristics. This derives from the finite repertoire of obtainable sounds and, more specifically, from the more easily producible dynamic variations associated with it. Thus, it is wise to use different techniques, the better to exploit their different potential. Moreover, the musician must study and experiment with a technique. This is essential in order to determine all its characteristics and to acquire a feeling for the parameter choices necessary for nontrivial use. In any case, a synthesis technique is simply a tool to produce sound, and sound is not yet music.

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BABBLE ONLINE: APPLYING STATISTICS AND DESIGN TO SONIFY THE INTERNET

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ABSTRACT

A statistician (Hansen) and a media artist (Rubin) investigate the application of statistical methods and sound-design principles to the real-time sonification of Internet communications. This paper presents results from two applications: the sonification of browsing activity on Lucent's Web site, and the sonification of a large

number of Internet chat sites in real-time. These experiments suggest new ways to experience the diverse and dynamic data streams generated by modern data networks. As an art-technology collaboration, the project outcomes range from the creation of art installations to the development of practical monitoring platforms. This paper discusses the interplay between these two perspectives, and suggests that each is motivated by a common interest in generating meaningful experiences with dynamic data.

1. INTRODUCTION

Modern work in sonification emerged in the literature on computer-human interfaces and over the last decade has matured into its own field of scientific inquiry. The use of sound in exploring the information hidden in data, the principles and broad application of auditory displays are eloquently described in [10]. Early application areas included real-time monitoring of financial data, medical diagnostics, and even air traffic control systems. Computer simulations also provided extensive data for sonification. Since that point, a virtual explosion has taken place in our ability to collect data relating to human communication and social systems.

Today, almost every aspect of our lives can be "rendered" digitally. Advances in data collection technologies have made commonplace continuous, high-resolution measurements of our physical environment (weather patterns, seismic events, ecological indicators). Equally open to observation are our routine movements through and interactions with our physical surroundings (automobile and air traffic, large-scale land use). In computer-mediated settings, our activities either depend crucially on or consist entirely of complex digital data (financial transactions, accesses to global information systems, Web site and Internet usage). As a reflection of the diversity and variety of the systems under study, these data-based descriptions of our daily lives tend to be massive in size, dynamic in character, and repleat with rich structures.

The advent of these enormous repositories of digital information presents us with an interesting challenge: How can we represent and interpret such complex, abstract and socially important data? In a new collaboration, Ear to the Ground [4], we have begun an exploration into ways of creating experiential encounters with otherwise abstract data streams, especially through sound.1 In [8], we discuss the broad goals of our collaboration and examine soni-

1 Ear to the Ground is part of the Arts in Multimedia project co-

fication from both an artistic as well as a data analytic perspective. In this paper, we examine the use of auditory displays in understanding large-scale Internet communications.

2. EXAMPLES

2.1. Web site traffic

Every day, large Web sites like Yahoo attract hundreds of thousands of visitors. During active periods, there can be thousands of people accessing data simultaneously from a Web site. For users of information portals like Yahoo, the speed of the servers (as reflected by rapid or sluggish responses) provides the only clue about the number of other people accessing information. While attempts have been made to visually assess browsing patterns in real-time [11, 1], the effectiveness of these displays deteriorates for high-traffic domains. For our first sonification example, we create an ambient display to characterize certain aspects of the activity on a busy Web site.

As you navigate the Web, your browser requests various kinds of data from one or more Web servers. As part of part of the process of delivering content, most Web servers will record information about the visitor and the items they requested. These items include HTML pages, images, Java class files and PostScript documents. The information available to the Web server about each request includes a timestamp, the IP address of the visitor's computer, the type of browser they are using, the URL of the requested item, the "referral page" (the URL that directed the visitor to the requested item), and perhaps a "cookie" to recognize returning visitors. These details are typically stored as a single line of a potentially enormous log file.2 The data for this experiment came from the Lucent Technologies corporate site, www.lucent.com. On a typical day, 60K visitors to this site will generate a 15Mb (compressed) log file, consisting of 700K entries. Given that our interest is on how users navigate the content of a site, we restrict our attention to HTML files, PostScript and PDF documents. All other requests made to the Web server are ignored, reducing the data by a factor of 10. We then further process the data to extract user paths or "visits," where a visit is a contiguous sequence of requests made by a user while browsing the site. Over 70% of the visitors to www.lucent.com look at just three pages or less, and hence a minority of the visits exhibit "interesting" navigational patterns.

sponsored by Lucent Technologies and the Brooklyn Academy of Music

(BAM). The authors gratefully acknowledge the help of project managers Wayne Ashley (BAM) and Marah Rosenberg (Lucent; now with Avaya Communications).
2 Each line is commonly referred to as a “hit.”

The Lucent Web site is built hierarchically, in the sense that pages deeper in the directory tree represent more detailed information than those at shallower levels. At its busiest, there can be as many as 300 people browsing www.lucent.com; while during the pre-dawn hours there can be as few as 5 simultaneous visitors. Our sonification is designed to convey qualitative information about site usage, answering questions like:

Overall, is the site busy or quiet?

What proportion of the visitors are delving for specific information deep within the site, as compared to those visitors who are “just passing through,” glancing briefly at the home page and then moving on?

How are users distributed across the various content areas of the site?

Which portions of the site are visited together? What kinds of patterns do we find in user behavior?

We think of this sonification as one possible “background” information stream that can inform content providers, Web designers and even the visitors themselves.

2.1.1. Sonification design

Our audio display makes use of the hierarchical structure of the content offered by www.lucent.com. First, a unique pitch was used to identify each of five high-level subdomains within the site: /micro, representing Lucent’s microelectronics design and manufacturing business (now Agere Systems); /enterprise, for the enterprise systems and software business (now Avaya Communications); /minds, a corporate introduction to Bell Labs research; /press, a collection of press releases and investor information; and /search, the local search engine for the site.

The total number of visitors accessing any information from a subdomain affects the loudness and tonal balance of a low-register drone at the associated pitch. Visitors requesting content deeper in the site are represented by higher-pitched pulsing tones (separated

by one or two octaves from the base pitch for the subdomain): the faster the pulse, the more people are accessing that area, and the greater the proportion of high-register sounds, the more detailed the content. By assigning well-separated pitches to each subdomain, shifts in activity both within and between the areas can be heard. In Table 1 we present a simple mapping of data collected by the Lucent Web server to a continuously time-varying vector of usage statistics. In the category of Overall browsing, we count any visitor accessing content pages (HTML, PostScript or PDF) from the indicated subdomain. A Mid-Level access is a request for content two or more directories down. Simple examples are /micro/K56flex/index.html (information on a brand of 56K modem) and /press/0101/010118.nsb.html (a press release for January 18, 2001). The final category, Deep browsing, refers to pages that are four or more directories down in the tree. One example is a paper from the April/June 2000 issue of the Bell Labs Technical Journal, located at /minds/techjournal/apr-jun2000/pdf/paper02.pdf.

Then, the resulting 15 values in Table 1, A1–E3, were mapped to sound as follows:

Overall activity Measured by A1–E1, voiced with a low-register drone. The aggregate number of visitors accessing information within each of the five areas modulates the loudness of each of the five pitches.

	/micro	/enterprise	/minds	/press	/search
Overall	A1	B1	C1	D1	E1
Mid-Level	A2	B2	C2	D2	E2
Deep	A3	B3	C3	D3	E3

Table 1: Mapping used for Web site traffic example. Overall activity records the movements of all users; Mid-Level counts users 2 or 3 directories into the site; Deep browsing consists of users 4+ directories down.

Mid-Level browsing Measured by A2–E2 and assigned a rhythmic middle-register tone pulse; pulse loudness and repetition speed rises and the timbral brightness increases as the volume of mid-level browsing increases. There are five independent pulses, each at a different fixed pitch, representing the five content areas.

Deep browsing Measured by A3–E3 and made audible via rhythmic high-register “ting” sounds (plucked steel string sam-

ples). Loudness and repetition speed rises as the volume of deep browsing increases. Again, there are five independent "ting" sounds, each at a different fixed pitch, representing the five content areas.

We used pitch groups that were consonant, and for the sounds that incorporated rhythm (A2–E3), the phase and frequency of each pulse in the matrix varies independently, yielding a sound with a changing rhythmic texture but no fixed beat.

The purpose of this sonification is to make interpretable the activities of users on a Web site. Therefore, the stream of hits being processed by a Web server (reduced to include only the HTML, PostScript and PDF documents) needs to be transformed to extract meaningful user-level data. A real-time monitoring tool was developed that maintains a bank of active visits (recording separately the activities of all the people browsing the site at a given time) and updates various statistics with each user request. When cookies or some other authentication mechanism allows us to recognize returning visitors, the monitor will update a more complicated user profile that encapsulates previous browsing patterns. Our traffic sonification as described above takes as input the location of each visitor within a site at a given point in time. When constructing more elaborate sound displays, our design will continue to focus on user activities, drawing more heavily on the statistics culled by the monitoring tool. This emphasis distinguishes our approach from sonification methods that assess Web server performance by making audible statistics relating to server load, HTTP errors, and agent types [?].

2.1.2. Impressions and extensions

We have created three audio examples for the activity on the Lucent site. Our data were captured on November 11, 1999 and we created sonifications of the traffic at 6:00 am, an extremely slow period for the site; noon, a relatively active time; and 2:30 pm, the point at which the site was busiest. The samples are located at our project Web site [6]. Even with this relatively straightforward mapping, one finds compelling patterns. For example, the affinity between the /enterprise subdomain and the /search facility can be heard as the pulses for these areas rise and fall together.[3]

[3] While clearly audible, these shifts can really only be precisely associated with areas after a certain amount of experience with the mapping.

Also, when comparing moderately active to extremely busy periods, we find that the number of people digging deep into the site is not a fixed fraction of the total number of visitors. That is, the volume of the low-register drones exhibits much more variation than the components for the other two categories of accesses. Each of these effects can be verified by examining the logs, reinforcing the usefulness of our sonification as a tool for constructing hypotheses about site traffic.

As mentioned at the beginning of this section, Web browsers offer a rich set of data about the visitor when requesting data from a server. This display makes use of only the most basic information about a visit, namely the depth of pages accessed. In ongoing work, we are augmenting our sonification with extra features derived both directly from the server data as well as from statistical navigation models [12] fit for the Web site under study. So far, we have found that such extensions are most effective when developed in the context of a particular monitoring application. For example, an extended version of this ambient display can aid system architects of large, Web hosting services understand cache performance and can aid in server provisioning. Another extension will make greater user of our navigation models and can help designers and usability engineers better architect Web sites. We will report on these and other developments through the project Web site [4].

2.2. Chat rooms and bulletin boards

At any given moment, tens of thousands of real-time conversations are taking place across the Internet on public forums, bulletin boards and chat sites. To imagine making these conversations simultaneously audible evokes an image of uproarious babble. And yet, in the aggregate, this massive stream of live communication could exhibit rich thematic structure. Can we find a meaningful way to listen in to so many conversations, rendering them in a way that is comprehensible and not overwhelming?

In some sense, a byproduct of our Web traffic sonification is the creation of a kind of community from the informal gathering of thousands of visitors to a given Web site. Traditionally, informational Web sites like www.lucent.com have provided us with very little sense of the other people who are requesting

data from the server. To attract and retain visitors, however, many commercial sites recognize the potential of the Web to form social as well as informational networks. As a result, Web-based forums, message boards and a variety of chat services are common components of current site designs. While Internet Relay Chat (IRC) has been a widely used standard since the inception of the Internet, the popularization of the Web has resulted in a virtual explosion of chat applications.4 For example, www.yahoo.com (a US-based Web portal) offers hundreds of separate chat rooms attracting tens of thousands of visitors a day. Specialized sites like www.style.com (the homepage for Vogue magazine) or www.audiworld.com (an resource for Audi owners) have also found their message boards to be the most frequently accessed parts of their domains.

To get a sense of the amount of content that is available in these dynamic formats, we examined sites contained in the DMOZ Open Directory [3], an open source listing of over 2 million Web sites compiled and categorized by 33,000 volunteer editors. From the November 20, 2000 image of the directory, we counted 36,681 separate sites offering some kind of chat, bulletin board or other public forum. While we did not examine the activity on all of these sites, the number is staggering. If we include other peer-to-peer communication technologies like instant messaging,5 the amount of dialogue taking place on the Web at any point in time is almost unfathomable. The goal of our second sonification is to make interpretable the thousands of streams of dynamic information being generated on the Web. In so doing, we attempt to characterize a global dialogue, integrating political debates, discussions of current events, and casual exchanges between members of virtual communities.

4 RC was developed by Jarkko Oikarinen in Finland in the late eighties, and was originally intended to work as a better substitute for talk on his bulletin board.

2.2.1. Content monitors and the statistics engine

Our starting point is text. Albeit diverse in style and dynamic in character, the text (or transcript) of these data sources carries their meaning. Therefore, any auditory display consisting only of generated tones would not be able to adequately represent the data without a very complex codebook. The design of our sonification then depends heavily on text-to-speech (TTS). As with the

traffic example in the previous section, we think of the audio output as another background information stream. The incorporation of spoken components in the sound design poses new challenges, both practical and aesthetic. For example, simply voicing every word taking place in a single chat room can produce too much text to be intelligible when played in real-time and can quickly exhaust the listener. Instead, we build a hierarchical representation of the text streams that relies on statistical processing for content organization and summarization prior to display.

Before considering sonification design, we first had to create specialized software agents that would both discover new chat rooms and message boards, as well as harvest the content posted to these sites. (See Figure 1 for an overview of our system architecture.) Most bulletin boards and some chat applications use standard HTML to store visitor contributions. In many cases, a specific login name is required to gain access to the site. For these situations, we constructed a content agent in Perl, as this language provides us the most convenient platform for managing access details (like cookies). The public chat rooms on sites like chat.yahoo.com can be monitored in this way. For IRC we built a configurable Java client that polls a particular server for active channels. Web sites like www.cnn.com (a popular news portal) and www.financialchat.com (a financial community hosting chat services for day traders) offer several IRC rooms, some of which are tightly moderated.

In addition to collecting content, each monitoring agent also summarizes the chat stream, identifying basic topics and updating statistics about the characteristics of the discussion: What percentage of visitors are contributing? How often to they contribute and at what length? Is the room "on topic," or are many visitors posting comments on very different subjects? Topics are derived from the chat stream using a variant of generalized sequence mining [7] that incorporates tags for the different parts of speech. While the exact details are beyond the scope of this abstract, a generalized sequence is a string of words possibly separated by a wildcard, "*". For example, if we let A, B and C denote specific "contentful" words (say, nouns, adjectives and adverbs), then AB C , A B C and A B C are all generalized sequences. The wildcard allows us to identify "Gore * disputes * election" from the sentences

5 AOL alone records tens of millions of people using their instant mes-

saging service each month.
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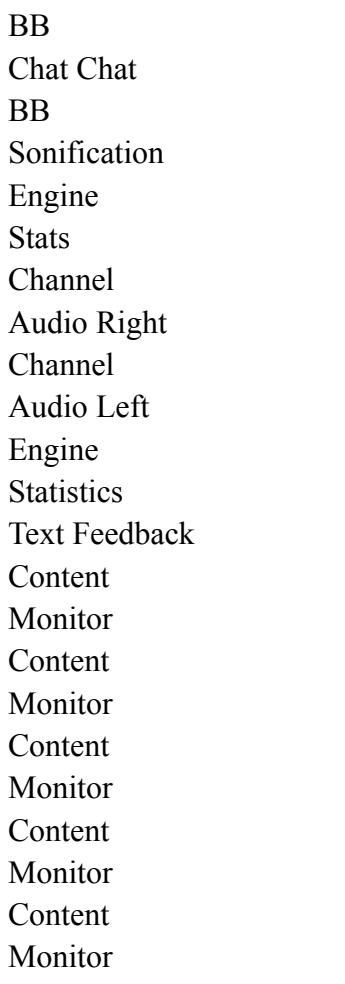
ICAD01-12


Figure 1: System architecture overview. A large number of content streams (Chat = chat rooms; BB = Bulletin boards) are gathered by specialized agents that transmit them in a homogenized format to the statistics engine. The statistics engine then distills the streams into a much smaller number of configurable text streams as well as a number of descriptive vectors. The sonification engine then "plays" these text and data streams. The entire systems operates in real-time.

"Vice President Gore filed papers to dispute the presidential election," "Aides for Gore indicated that he has every reason to dispute the election", and "Gore is still deciding whether or not to dispute the election".

As many posts to chat rooms contain spelling mistakes and incorrect grammar, assigning words to different parts of speech is error-prone. However, unlike most applications of statistical natural language processing, our content monitors update their summaries each time new material is posted and downweight older contributions. Because our sonification renders these sources in real-time, small mistakes have little effect on the power of the overall display to convey the ideas being discussed.

Each of the content monitors are periodically polled by the statistics engine (see Figure 1). This Java-application clusters the different chat rooms and bulletin boards based on their topic and numerical summaries. As the topic in a room changes over time, the statistics engine is constantly updating and reformulating cluster membership. Because a content stream can in fact support a number of simultaneous discussions (the threads of a bulletin board, say), we employ a soft-clustering technique. In our initial work, we have used a mixture-based scheme that determines the number of clusters with an MDL (Minimum Description Length) criterion [9]. Each room is then assigned a probability that it belongs to the different groups. This model also provides for topic summarization at the cluster-level. Next, a stochastic framework was developed to sample representative sentences posted to the chat or bulletin board. When a discussion is extremely unstructured, this selection is essentially random sampling from all the contributions added to the chat since the last polling point. In addition to textual data streams, the statistics engine is also responsible for communicating the various ingredients for the display to our sonification engine, Max/MSP [2] (see Figure 1). We have adopted the Open Sound Control [13] protocol from Center for New Music and Audio Technologies to transfer data between the statistics engine (running on a Macintosh with LinuxPPC) and the sonification engine (running on a Macintosh with OS/9).

2.2.2. Sonification design

As with the previous example (Section 2), our goal is to create a sonification that is both communicative and listenable. Here we face the additional challenge of incorporating verbal content. With TTS annotations, it becomes more difficult to intelligibly convey more than one layer of information through the audio channel. Our design incorporates spatialization, pitch and timbral differentiation, and rhythm to achieve clarity in the presentation of the hi-

erarchically structured data coming from the statistics engine. The auditory display cycles through topic clusters, spending relatively more time on subjects being actively discussed by the largest numbers of people. Each different topic is assigned a different pitch group, reinforcing subject changes when they occur. For each cluster, the statistics engine sends three streams of information to the sonification engine:

Topics A continuously updated list of up to ten "topics" (the most frequently appearing words and phrases – generalized sequences – mined from the multiple chat streams associated with the given cluster; the number of topics is configurable, but ten was chosen based on timing considerations);

Content samples A selection of sample sentences, identified by the statistics engine as typical or representative, in which these topics appear;

Content entropy A vector that represents the changing level of entropy in the source data.

The topics are spoken by the TTS system6 at regular intervals in a pitched monotone, and are panned alternately hard left and hard right in the stereo field, creating a sort of rhythmic "call and response." The sample sentences are panned center, and rendered with limited inflection (as opposed to the pitched monotone of the topics). The tonal, rhythmic and spatial qualities of the topics contrasts sufficiently with the sample sentences to create two distinctly comprehensible streams of verbal information.

The entropy vector controls an algorithmic piano score. When entropy is minimal and the discussion in the chat room or bulletin board is very focused on one subject, chords are played rhythmically in time with the rhythmic recitation of the topics. As entropy increases and the conversations diverge, a Gaussian distribution is used to expand the number, range and dynamics of notes that fall between the chords. With this audio component, one can easily differentiate a well-moderated content source from a more free-form, public chat without distracting from the TTS annotations. The piano score also serves a secondary function as an accompaniment to the vocal foreground, enhancing the compositional balance and overall musicality of the sound design.

2.2.3. Sample sonification and impressions

On our project Web site [5], we have a sample chat room sonification that cycles through three topics. In this sound file, we are

6 The built-in MacOS TTS capability controlled by Max/MSP.

listening to the output of only three content monitors. Hence, by design, each topic is confined to a single site. The first portion of this example (ending at 1:47 into the sample) concerns the recent recall of Bridgestone tires and was based on a www.cnn.com chat room. This discussion was heavily moderated and hence the backing piano score frequently reduces to a simple rhythm. For our second topic (from 1:47 to 3:21 of the sample) we recorded chat exchanges on www.financialchat.com one morning when Yahoo's stock opened low. In this example, we hear day traders frantically exchanging predictions about when Yahoo's stock will "bounce." The final topic in this sample (from 3:21 to the end) is again from www.cnn.com and treats a recent strike by the Screen Actor's Guild and the American Federation of Television and Radio Artists. This chat room was much less moderated than the previous CNN chat, and the backing piano score reflects that. Although this example does not make full use of the clustering capabilities of the statistics engine, the essence of our sonification design is clear. The audio display provides an informative and accessible representation of dynamic, textual content. The topic and content sample streams are easy to separate, and when placed in the background, call our attention to important new subjects being discussed on the Web.

2.2.4. Applications and Extensions

Our sonification provides an audible interface to the (now) massive amount of dynamic content available on the Web. Given the preprocessing that takes place in the content monitors and the statistics engine, a simple extension is to provide search-like functionality. A user can register interest in a certain topic and "tune" our display to present only rooms where this subject is being discussed. The necessary ingredients to implement this feature are all currently available in the statistics engine. Similarly, one can easily restrict the sites that are used for the display. When a new subject appears that draws the user's interest, it is also trivial to add a feature that would direct the user's browser to one or more chats associated with the topic. As a final extension, we have provided the content monitors with a configurable list of Web sites

that can be used to help disambiguate elements in the chat stream. For example, the day traders speak in ticker symbols. Providing the content monitor with the URL for the ticker symbol look-up service offered by Yahoo allows the content monitor to weave not only company names but also recent company-related headlines directly into the stream fed to the statistics engine.

While we have focused mainly on chat and bulletin boards, this technology can be applied in other settings. We have begun collaborating with the designers of a natural language interface for Web-based help systems. Here, we give voice to the hundreds of simultaneous conversations taking place between Web site visitors and the automated help system. A similar display can be imagined for other natural language interfaces, including search engines like AskJeeves (www.jeeves.com). In general, the practical applications of this summarization and auditory display tool abound.

3. CONCLUSION AND COMMENTS ON COLLABORATIVE RESEARCH

The two applications outlined in this paper are the first outcomes of a collaboration sponsored by Bell Laboratories and the Brooklyn Academy of Music under the Arts in Multimedia project (AIM). The goal of AIM is to bring together researchers (in this case a statistician) and artists (in this case a sound artist), with the objective of advancing our separate agendas through collaborative projects. Our work together is predicated on the notion that sophistication both in data treatment and aesthetics are crucial to the successful design of audio displays. Thus, in each of our examples, we have endeavored to create a result which communicates information clearly, yet at the same time sounds well composed and appealing. Moving forward, it is our intention to apply these techniques both to practical applications, and also to create a series of artworks. These artworks will use our sonification techniques to establish a series of real-time listening posts, both on the Web and in physical locations. The listening posts will tap in to various points of interest on the Internet, using sound to reveal patterns and trends that would otherwise remain hidden.

In terms of applications, we are exploring the use of sonification to support the design, provisioning and monitoring of communication networks. A network operations center (NOC), for example, routinely receives clues about the health of the system in the form of text messages generated by routers and switches. An audio

display installed inside a NOC can act as an early warning system for approaching bottlenecks as well as aid in troubleshooting. By continued exposure to the sound of a "normally" functioning network, operators will be alerted to system changes that could signal problems.

Art emerges unexpectedly from experimentations with new statistical methods or considerations involving practical applications; and new tools for data analysis and modeling develop in response to artistic concerns. Each of us continues to be surprised by the connections that emerge from rethinking familiar problems in a new context. Through our project, we hope to illustrate both the value of art-technology collaborations as well as their necessity, especially when finding meaning in complex data.

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TAXONOMY AND DEFINITIONS FOR SONIFICATION AND AUDITORY DISPLAY

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ABSTRACT

Sonification is still a relatively young research field and many terms such as sonification, auditory display, auralization, audification have been used without a precise definition. Recent developments such as the introduction of Model-Based Sonification, the establishment of interactive sonification and the increased interest in sonification from arts have raised the need to revisit the definitions in order to move towards a clearer terminology. This paper introduces a new definition for sonification and auditory display that emphasizes the necessary and sufficient conditions for organized sound to be called sonification. It furthermore suggests a taxonomy, and discusses the relation between visualization and sonification. A hierarchy of closed-loop interactions is furthermore introduced. This paper aims to initiate vivid discussion towards the establishment of a deeper theory of sonification and auditory display.

1. INTRODUCTION

Auditory Display is still a young research field whose birth may be perhaps best traced back to the first ICAD conference1 in 1992 organized by Kramer. The resulting proceedings volume “Auditory Display” [1] is still one of the most important books in the field. Since then a vast growth of interest, research, and initiatives in auditory display and sonification has occurred. The potential of sound to support human activity, communication with technical systems and to explore complex data has been acknowledged [2] and the field has been established and has clearly left its infancy.
As in every new scientific field, the initial use of terms

lacks coherence and terms are being used with diffuse definitions. As the field matures and new techniques are discovered, old definitions may appear too narrow, or, in light of interdisciplinary applications, too unspecific. This is what motivates the redefinitions in this article.

The shortest accepted definition for sonification is from Barrass and Kramer et al. [2]: “Sonification is the use of non-speech audio to convey information”. This definition excludes speech as this was the primary association in the

1see www.icad.org

auditory display of information at that time. The definition is unclear about what is meant by conveyance of information: are real-world interaction sounds sonifications, e.g. of the properties of an object that is being hit? Is a computer necessary for its rendition? As a more specific definition, the definition in [2] continues:

> “Sonification is the transformation of data relations into perceived relations in an acoustic signal for the purposes of facilitating communication or interpretation.”

It is significant that the emphasis here is put on the purpose of the usage of sound. This automatically distinguishes sonification from music, where the purpose is not on the precise perception of what interactions are done with an instrument or what data caused the sound, but on an underlying artistic level that operates on a different level. Often, the word ‘mapping’ has been used interchangeably with ‘transformation’ in the above definition. This, however, suggests a severe limitation of sonification towards just mappings between data and sound – which was perfectly fine at the time of the definition where such a ‘Parameter-Mapping Sonification’ was the dominating paradigm.

However, the introduction of Model-Based Sonification (MBS) [3, 4] demonstrates methods to explore data by using sound in a way that is very different from a mapping: in Parameter-Mapping Sonification, data values are mapped to acoustic attributes of a sound (in other words: the data ‘play’ an instrument), whereas in MBS sonification models create and configure dynamic processes that do not make sound at all without external interactions (in other words:

the data is used to build an instrument or sound-capable object, while the playing is left to the user). The user excites the sonification model and receives acoustic responses that are determined by the temporal evolution of the model. By doing this, structural information is holistically encoded into the sound signal, and is no longer a mere mapping of data to sound. One can perhaps state that data are mapped to the configurations of sound-capable objects, but not that they are mapped to sound.

Clearly, sonification models implemented according to MBS are very much in line with the original idea that sonifi-

cation allows for the discovery of structures in data through sound. Therefore there is the need to reformulate or adapt the definition for sonification to better include such uses of sound, and beyond that hopefully other possible yet-to-be-discovered linkages between data and sound.

Another challenge for the definition comes from the use of sonification in the arts and music: recently more and more artists incorporate methods from sonification in their work. What implications does this have for the term sonification? Think of scientific visualization vs. art: what is the difference between a painting and a modern visualization? Both are certainly organized colors on a surface, both may have aesthetic qualities, yet they operate on a completely different level: the painting is viewed for different layers of interpretation than the visualization. The visualization is expected to have a precise connection to the underlying data, else it would be useless for the process of interpreting the data. In viewing the painting, however, the focus is set more on whether the observer is being touched by it or what interpretation the painter wants to inspire than what can be learnt about the underlying data. Analogies between sonification and music are close-by.

Although music and sonification are both organized sound, and sonifications can sound like music and vice versa, and certainly sonifications can be ‘heard as’ music as pointed out in [5], there are important differences which

are so far not manifest in the definition of sonification.

2. A DEFINITION FOR SONIFICATION

This section introduces a definition for sonification in light of the aforementioned problems. The definition has been refined thanks to many fruitful discussions with colleagues as listed in the acknowledgements and shall be regarded as a new working definition to foster ongoing discussion in the community towards a solid terminology.

Definition: A technique that uses data as input, and generates sound signals (eventually in response to optional additional excitation or triggering) may be called sonification, if and only if

(C1) The sound reflects objective properties or relations in the input data.

(C2) The transformation is systematic. This means that there is a precise definition provided of how the data (and optional interactions) cause the sound to change.

(C3) The sonification is reproducible: given the same data and identical interactions (or triggers) the resulting sound has to be structurally identical.

(C4) The system can intentionally be used with different data, and also be used in repetition with the same data.Data Sonification

Algorithm

systematic

transformation reproducable exchangeability

of data

interactions (optional)

Definition: Sonification

Figure 1: Illustration of the general structure and necessary conditions for sonification. The yellow box depicts besides the sonification elements few other components of auditory displays, see also Sec. 3.

This definition emphasizes important prerequisites for the scientific utility of sonification. It has several partly unexpected implications that are to be explored in the following discussion.

2.1. Discussion

2.1.1. General Comments

Sonification Techniques: According to the above defini-

tion, the techniques Audification, Earcons, Auditory Icons, Parameter-Mapping Sonification as well as Model-Based Sonification are all covered by the definition – they all represent information/data by using sound in an organized and well-structured way and they are therefore different sonification technique.2 This may first appear unfamiliar in light of the common parlance to see earcons/auditory icons as different from sonification. However, imagine an auditory display for biomedical data that uses auditory icons as sonic events to represent different classes (e.g. auditory icons for benign/malignant tissue). The sonification would then be the superposition or mixture of all the auditory icons chosen for instance according to the class label and organized properly on the time axis. If we sonify a data set consisting only of a single data item we naturally obtain as an extreme case a single auditory icon. The same can be said for earcons. Although sonification originally has the connotation of representing large and complex data sets, it makes sense for the definition to also work for single data points.

Data vs. Information: A distinction between data and information is – as far as the above definition – irrelevant. Think of earcons to represent computer desktop interactions such as “delete file”, “rename folder”. There can be a lexi-

2they are also covered by the definition of sonification as ‘non-speech use of sound to convey information’!

con of terms (file, folder, link) and actions (delete, rename, etc.), and in practical computer implementations these features would be represented numerically, e.g. object = O1, action = A3. By doing so, the information has been turned into data, and this is generally done if there is more than one signal type to give. Information like for instance a verbal message can always be represented numerically and thus be understood as data. On the other side, raw data values often carry semantic interpretation: e.g. the outside temperature data value -10◦C (a one-dimensional data set of size 1) – this is cold, and clearly information! Assuming that information is always encoded as data values for

its processing we can deal with both in a single definition. How the data are then represented by using sound is another question: whether sonification techniques use a more symbolic or analogic representation according to the analogic-symbolic continuum of Kramer [6] is secondary for the definition.

Mapping as a specific case of sonification: Some articles have used "sonification" to refer specifically to mapping-based sonification, where data features are mapped to acoustic features of sound events or streams. Yet sonification is more generally the representation of data by using sound. There may be times when a clear specification of the sonification technique, e.g. as model-based, audification or parameter-mapping sonification, may be helpful to avoid confusion with the general term of sonification. It makes sense to always use the most specific term possible, that is to use the term Parameter Mapping Sonification, Audification, Model-Based Sonification, etc. to convey exactly what is meant. The term Sonification, however, is, according to the definition, more general which is also supported by many online definitions3. In result we suggest using sonification with the same level of generality as the term visualization is used in visual display.

Sonification as algorithm and sound: Sonification refers to the technique and the process, so basically it refers to the algorithm that is at work between the data, the user and the resulting sound. Often, and with equal right, the resulting sounds are called sonifications. Algorithm means a set of clear rules, independent of whether it is implemented on a computer or any other way.

Sonification as scientific method: According to the definition, sonification is an accurate scientific method which leads to reproducible results, addressing the ear rather then the eye (as visualization does). This does not limit the use of sonifications to data from the sciences, but only states that sonification can be used as a valid instrument to gain insight. The subjectivity in human percep-

3http://en.wikipedia.org/wiki/Sonification,
http://wvvel.csee.wvu.edu/sepscor/sonification/lesson9.html,
http://www.techfak.uni-bielefeld.de/ags/ni/projects/datamining/datason/

datason e.html, http://www.cs.uiowa.edu/ kearney/22c296Fall02/ CrittendonSpecialty.pdf, to name a few.

tion and interpretation is shared with other perceptualization techniques that bridge the gap between data and the human sensory system. Being a scientific method, a prefix like in "scientific sonification" is not necessary.

Same as some data visualizations may be 'viewed' as art, sonifications may be heard as 'music'[5], yet this use differs from the original intent.

2.1.2. Comments to (C1)

(C1) The sound reflects objective properties or relations in the input data.

Real-world acoustics are typically not a sonification although they often deliver object-property-specific systematic sound, since there is no external input data as requested in C1. For instance, with a bursting bottle, one can identify what is the data, the model and the sound, but the process cannot be repeated with the same bottle. However, using a bottle that fills with rain, hitting it with a spoon once a minute can be seen as a sonification: The data here is the amount of rainfall, which is here measured by the fill level, and the other conditions are also fulfilled. Tuning a guitar string might also be regarded as a sonification to adjust the tension of a string4. These examples show that sonifications are not limited to computer-implementations according to the definition, which embraces the possibility of other non-computer-implemented sonifications.

The borders of sonification and real-world acoustics are fuzzy. It might be discussed how helpful it is to regard or denote everyday sounds as sonifications.

2.1.3. Comments to (C2)

(C2) The transformation is systematic. This means that there is a precise definition provided of how the data (and optional interactions) cause the sound to change.

What exactly do we mean by "precise"? Some sound generators use noise and thereby random elements so that sound events will per se sound different on each rendering. In Parameter-Mapping Sonifications, the intentional addition of noise (for instance as onset jitter to increase per-

ceptability of events that would otherwise coincide) is often used and makes sense. In order to include such cases randomness is allowed in the definition, yet it is important to declare where and what random elements are used (e.g. by describing the noise distribution). It is also helpful to give a motivation for the use of such random elements. By using too much noise, it is possible to generate useless sonifications in the sense that they garble interpretation of the underlying data. In the same way it is possible to create useless scientific visualizations.

4thanks to the referee for this example!

2.1.4. Comments to (C3)

(C3) The sonification is reproducible: given the same data and identical interactions (or triggers) the resulting sound has to be structurally identical.

The definition claims reproducibility. This may not strictly be achieved for several reasons: the loudspeakers may generate a different sound at different temperatures, other factors such as introduced noise as discussed above may have been added. The use of the term “structurally identical” in the definition aims to weaken the stronger claim of sample-based identity. Sample-based identity is not necessary, yet all possible psychophysical tests should come to identical conclusions.

2.1.5. Comments to (C4)

(C4) The system can intentionally be used with different data, and also be used in repetition with the same data.

Repeatability is essential for a technique to be scientifically valid and useful – otherwise nobody could check the results obtained by using sonification as instrument to gain insight. However, there are some implications by claiming repeatability for what can and cannot be called sonification. It has for instance been suggested that a musician improvising on his instrument produces ‘a sonification of the musician’s emotional state’. With C4, however, “play-

ing a musical instrument" is not a sonification of the performer's emotional state, since it can not be repeated with the 'identical' data. However, the resulting sound may be called a sonification of the interactions with the instrument (regarded here as data), and in fact, music can be heard with the focus to understand the systematic interaction patterns with the instruments.

Some of these conditions have been set as constraints for sonification, e.g. reproducibility in the 'Listening to the Mind Listening' concert5, but not been connected to a definition of sonification.

In summary, the given definition provides a set of necessary conditions for systems and methods to be called sonification. The definition is neither exhaustive nor complete; we hope it will serve as the core definition as we as community work towards a complete one.

3. SONIFICATION AND AUDITORY DISPLAY

With the above definition, the term sonification takes the role of a general term to express the method of rendering sound in an organized and well-structured way. This is in good analogy with the term visualization which is also the general term under which a variety of specific techniques such as bar charts, scatter plots, graphs, etc. are subsumed. Particularly there is an analogy between scatter plots where graphical symbols (data-mapped color/size...) are organized in space to deliver the visualization, and Parameter-Mapping Sonification, where in a structurally identical way acoustic events (with data-mapped features) are organized in time. It is helpful to have with sonification a term that operates on the same level of generality as visualization.

This raises the question what then do we mean by auditory displays? Interestingly, in the visual realm, the term 'display' suggests a necessary but complementary part of the interface chain: the device to generate structured light/images, for instance a CRT or LCD display or a projector. So in visualization, the term visualization emphasizes the way how data are rendered as an image while the display is necessary for a user to actually see the information. For auditory display, we suggest to include this aspect of con-

5http://www.icad.org/websiteV2.0/Conferences/ICAD2004/concert call.htm

version of sound signals into audible sound, so that an auditory display encompasses also the technical system used to create sound waves, or more general: all possible transmissions which finally lead to audible perceptions for the user. This could range from loudspeakers over headphones to bone conduction devices. We suggest furthermore that auditory display should also include the user context (user, task, background sound, constraints) and the application context, since these are all quite essential for the design and implementation. Sonification is thereby an integral component within an auditory display system which addresses the actual rendering of sound signals which in turn depend on the data and optional interactions, as illustrated in Fig. 2.

Figure 2: Auditory Displays: systems that employ sonification for structuring sound and furthermore include the transmission chain leading to audible perceptions and the application context.

tion since for instance dialogue systems and speech interfaces may also be regarded as auditory displays since they use sound for communication. While such interfaces are not

the primary focus in this research field the terminology suggests their inclusion. On the other hand, Auditory Display may be seen as a subset of the more general term of Auditory Interfaces which do not only include output interfaces (auditory displays, sonification) but also auditory input interfaces which engender bidirectional auditory control and communication between a user and a (in most cases) technical system (e.g. voice control system, query-by humming systems, etc.).

4. HIERARCHY FROM SOUND TO SONIFICATION

So far we have dealt with the necessary conditions surrounding sonification and thus narrowed sonification down to a specific subset of using sound. In this section, we look at sonification in a systemic manner to elucidate its superordinate categories. Figure 3 shows how we suggest to organize the different classes of sound. On the highest level,Map of Sound
Organized Sound
Functional Sounds
Music &
Media Arts Sonification(a)
(b)
Figure 3: Systemic map of sound, showing sonification and its relation to other categories.
sounds are here classified as Organized Sound and unorganized sound. Organized sounds separate from random or otherwise complex structured sounds in the fact that their occurence and structure is shaped by intention. Environmental sounds appear often to be very structured and could thus also be organized sounds, however, if so, any sound would match that category to some extent. It thus may be useful to apply the term to sounds that are intentionally organized – in most cases by the sound/interface developer.
The set of organized sound comprises two large sets that partially overlap: music and functional sounds. Music is without question a complex structured signal, organized on various levels, from the acoustic signal to its temporal organization in bars, motifs, parts, layers. It is not our purpose to give a definition of music.
The second set is functional sounds. These are orga-

nized sounds that serve a certain function or goal [7]. The function is the motivation for their creation and use. To give an example, all signal sounds (such as telephones, doorbells, horns and warning hooters) are functional sounds. Certainly there are intersections with music, as music can serve functional aspects. For instance, trombones and kettle drums have been used to demonstrate kingship and power. A more subtle function is the use of music in supermarkets to enhance the 'shopping mood'. For that reason these sets overlap – the size of the overlap depends on what is regarded as function.

Sonification in the sense of the above definition is certainly a subset of functional sounds. The sounds are rendered to fulfill a certain function, be it communication of information (signals & alarms), the monitoring of processes, or to support better understanding of structure in data under analysis. So is there a difference between functional sounds and sonification at all? The following example makes clear that sonification is really a subset: Recently a new selective acoustic weapon has been used, the mosquito device6, a loudspeaker that produces a HF-sound inaudible to older people, which drives away teenagers hanging around in front of shops. This sound is surely functional, yet it could neither pass as sonification nor as music.

Finally, we discuss whether sonification has an intersection with music&media arts. Obviously there are many examples where data are used to drive aspects of musical performances, e.g. data collected from motion tracking or biosensors attached to a performer. This is, concerning the involved techniques and implementations similar to mapping sonifications. However, a closer look at our proposed definition shows that often the condition for the transformation to be systematic C2 is violated and the exact rules are not made explicit. But without making the relationship explicit, the listener cannot use the sound to understand the underlying data better. In addition, condition C4 may often be violated. If sonification-like techniques are employed to obtain a specific musical or acoustic effect without transparency between the used data and details of the sonification techniques, it might, for the sake of clarity, better be denoted

as ‘data-inspired music’, or ‘data-controlled music’ than as sonification. Iannis Xenakis, for instance, did not even want the listener to be aware of the data source nor the rules of sound generation.

6see http://www.compoundsecurity.co.uk/, last seen 2008-01-16

5. CLOSED INTERACTION LOOPS IN AUDITORY DISPLAYS

This section emphasizes the role of interaction in sonification. We propose different terms depending on the scope of the closure of the interaction loop. The motivation for this discussion is that it might be helpful to address how terms such as biofeedback or interactive sonification relate to each other.

We start the discussion with Fig. 4 that depicts closed loop interactions. The sonification module in the upper center playing rendered sonifications to the user. Data sources for sonification enter the box on the left side and the most important parts are (a) World/System: this comprises any system in the world that is connected to the sonification module, e.g. via sensors that measure its state, and (b) Data: these are any data under analysis or represented information to be displayed that are stored separately and accessible by the sonification.World/System
Sonification
Interactive Sonification
Human Activity (supported by sonification)
Auditory Biofeedback
Data
Navigation
Monitoring
No Action

Figure 4: Illustration of Closed-Loop Auditory Systems.

In this setting, Process Monitoring is the least interactive sonification, where data recorded from the world (in real-time) or read from the data repository is continuously used as input for a sonification rendering process. Here, the listener is merely passively listening to the sound with the

only active component being his/her focus of attention onto parts of the sound. Certainly, certain changes in the sound might attract attention and force the user to act (e.g. sell stocks, stop a machine, etc...).

A higher degree of active involvement occurs when the user actively changes and adjusts parameters of the sonification module, or interacts otherwise with the sonification system. We denote this case as Interactive Sonification. There is a wide field of possibilities of why and how to do so, and we discuss 3 different prototypical examples:

(a) Triggering: Consider a mapping sonification of a given data set. An essential interaction for the user is to issue the command to render/playback the sonification for a selected dataset. Possibly he/she does this several times in order to attend different parts of the sound signal. This elementary case is an interaction, however, a very basic one.

(b) Parameter Adjustment is done when the user changes parameters, such as what data feature are mapped to acoustic parameters, control ranges, compression factors, etc. Often such adjustments happen separate from the playback so that the changes are made and afterwards the updated sound is rendered. However, interactive real-time control is feasible in many cases and shows a higher degree of interactivity. The user actively explores the data by generating different 'views' of the data [8]. In visualization a similar interactivity is obtained by allowing the user to select axes scalings, etc.

(c) Excitatory Interaction is the third sort of interaction and is structurally similar to the case of triggering. Particularly in Model-Based Sonification [4], usually the data are used to configure a sound-capable virtual object that in turn reacts on excitatory interactions with acoustic responses whereby the user can explore the data interactively. Excitation puts energy into the dynamic system and thus initiates an audible dynamical system behavior. Beyond a simple triggering, excitatory interactions can be designed to make use of the fine-grained manipulation skills that human

hands allow, e.g. by enabling to shake, squeeze, tilt or deform the virtual object, for instance using sensor-equipped physical interfaces to interact with the sonification model. A good example for MBS is Shoogle by Williamson et al. [9], where short text messages in a mobile phone can be overviewed by shaking a mobile phone equipped with accelerometer sensors, resulting in audible responses of the text messages as objects moving virtually inside the phone. Excitatory interactions offer rich and complex interactions for interactive sonification.

The next possibility for a closed loop is by interactions that select or browse data. Since data are chosen, it may best be referred to as Navigation. Navigation can also be regarded as special case of Interactive Sonification, depending on where the data are selected and the borders are here really soft. Navigation usually goes hand in hand with triggering of sonification (explained above).

Auditory Biofeedback can be interpreted as a sonification of measured sensor data. In contrast to the above types, the user's activity is not controlling an otherwise autonomous sonification with independent data, but it produces the input data for the sonification system. The user perceives a sound that depends on his/her own activity. Such systems have applications that range from rehabilitation training to movement training in sports, e.g. to perform

a complex motion sequence (e.g. a tennis serve) so that its sonification is structurally more similar to the sonification of an expert performing the action [10].

The final category is Human Activity, which means that the interaction ranges beyond the sonification system into the world, often driven by the goal to change a world state in a specific way. In turn, any sensors that pick up the change may lead to changes in the sonification. The difference between the loop types before is that the primary focus is to achieve a goal beyond the sonification system, and not to interact with a closed-loop sonification system. Even

without attending the sonification consciously or primarily, the sound can be helpful to reach the goal. For example, imagine the real-world task to fill a thermos bottle with tea. While your primary goal is to get the bottle filled you will receive the 'gluck-gluck' sound with increasing pitch as a by-product of the interaction. If this is consistently useful, you subconsciously adapt your activity to exploit the cues in the sound – but the sound is only periphery for the goal. In a similar sense, sonifications may deliver helpful by-products to actions that change the world state. We regard such interaction add-ons where sonification is a non-obtrusive yet helpful cue for goal attainment as inspiring design direction. Such sonifications might even become subliminal in the sense that users, when asked about the sound, are not even aware of the sound, yet they perform better with sound than without.

6. DISCUSSION AND CONCLUSION

The definitions in this paper are given on the basis of three goals: (i) to anchor sonification as a precise scientific method so that it delivers reproducible results and thus can be used and trusted as instrument to obtain insight into data under analysis. (ii) to offer a generalization which does not limit itself to the special case of mappings from data to sound, but which introduces sonification as general systematic mediator between data and sound, whatever the representation might be. (iii) to balance the definition so that the often-seen pair of terms 'visualization & sonification' are at the same level of generality.

The definition has several implications which have been discussed in Sec. 2. We'd like to emphasize that this effort is being done in hope that the definition inspires a general discussion on the terminology and taxonomy of the research field of auditory display. An online version of the definition is provided at www.sonification.de with the aim to collect comments and examples of sonifications as well as examples that are agreed not to be sonifications and which help in turn to improve the definition.

In Section 3, we described integral parts for auditory display so that sonification takes a key component as the technical part involving the rendition of sound. Again, the

suggested modules are meant as working hypothesis to be discussed at ICAD.

While the given definitions specified terms on a horizontal level, Section 4 proposes a vertical organization of sound in relation to often used terms. The intersections between the different terms and categories have been addressed with examples.

Finally, we have presented in Section 5 an integrative scheme for organizing different classes of auditory closed loops according to the loop closure scope. It proves helpful to clarify classes of interactive sonifications. We think that grouping existing sonifications according to these categories can be helpful to better find alternative approaches for a given task.

The suggested terminology and taxonomy is the result of many discussions and a thorough search for helpful concepts. We suggest it as working definitions to be discussed at the interdisciplinary level of ICAD in hope to contribute towards a maturing of the fields of auditory display and sonification.

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Nomadic Radio: Scaleable and Contextual Notification for Wearable Audio Messaging.
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Nomadic Radio: Scaleable and Contextual Notification for Wearable Audio Messaging
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ABSTRACT
Mobile workers need seamless access to communication and information services on portable devices. However

current solutions overwhelm users with intrusive and ambiguous notifications. In this paper, we describe scaleable auditory techniques and a contextual notification model for providing timely information, while minimizing interruptions. User's actions influence local adaptation in the model. These techniques are demonstrated in Nomadic Radio, an audio-only wearable computing platform.
Keywords
Auditory I/O, passive awareness, wearable computing, adaptive interfaces, interruptions, notifications
INTRODUCTION
In today's information-rich environments, people use a number of appliances and portable devices for a variety of tasks in the home, workplace and on the run. Such devices are ubiquitous and each plays a unique functional role in a user's lifestyle. To be effective, these devices need to notify users of changes in their functional state, incoming messages or exceptional conditions. In a typical office environment, the user attends to a plethora of devices with notifications such as calls on telephones, asynchronous messages on pagers, email notification on desktop computers, and reminders on personal organizers or watches. This scenario poses a number of key problems.

Lack of Differentiation in Notification Cues
Every device provides some unique form of notification. In many cases, these are distinct auditory cues. Yet, most cues are generally binary in nature, i.e. they convey only the occurrence of a notification and not its urgency or dynamic state. This prevents users from making timely decisions about received messages without having to shift focus of attention (from the primary task) to interact with the device and access the relevant information.

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Minimal Awareness of the User and Environment

Such notifications occur without any regard to the user's engagement in her current activity or her focus of attention. This interrupts a conversation or causes an annoying disruption in the user's task and flow of thoughts. To prevent undue embarrassment in social environments, users typically turn off cell-phones and pagers in meetings or lectures. This prevents the user from getting notification of timely messages and frustrates people trying to get in touch with her.

No Learning from Prior Interactions with User

Such systems typically have no mechanism to adapt their behavior based on the positive or negative actions of the user. Pagers continue to buzz and cell-phones do not stop ringing despite the fact that the user may be in a conversation and ignoring the device for some time.

Lack of Coordinated Notifications

All devices compete for a user's undivided attention without any coordination and synchronization of their notifications. If two or more notifications occur within a short time of each other, the user gets confused or frustrated. As people start carrying around many such portable devices, frequent and uncoordinated interruptions inhibit their daily tasks and interactions in social environments.

Given these problems, most devices fail to serve their intended purpose of notification or communication, and thus do not operate in an efficient manner for a majority of their life cycle. New users choose not to adopt such technologies, having observed the obvious problems encountered with their usage. In addition, current users tend to turn off the devices in many situations, inhibiting the optimal operation of such personal devices.

Nature of Interruptions in the Workplace

A recent observational study [4] evaluated the effect of interruptions on the activity of mobile professionals in their workplace. An interruption, defined as an asynchronous and unscheduled interaction, not initiated by the user, results in the recipient discontinuing the current activity. The results revealed several key issues. On average, sub.jects were interrupted over 4 times per hour, for an average duration

slightly over 2 minutes. Hence, nearly 10 minutes per hour

was spent on interruptions. Although a majority of the interruptions occurred in a face-to-face setting, 20% were due to telephone calls (no email or pager activity was analyzed in this study). In 64% of the interruptions, the recipient received some benefit from the interaction. This suggests that a blanket approach to prevent interruptions, such as holding all calls at certain times of the day, would prevent beneficial interactions from occurring. However in 41% of the interruptions, the recipients did not resume the work they were doing prior to it. But active use of new communication technologies makes users easily vulnerable to undesirable interruptions.

These interruptions constitute a significant problem for mobile professionals using tools such as pagers, cell-phones and PDAs, by disrupting their time-critical activities. Improved synchronous access using these tools benefits initiators but leaves recipients with little control over the interactions. The study suggests development of improved filtering techniques that are especially light-weight, i.e. don't require more attention from the user and are less disruptive than the interruption itself. By moving interruptions to asynchronous media, messages can be stored for retrieval and delivery at more appropriate times.

NOMADIC RADIO: WEARABLE AUDIO MESSAGING

Personal messaging and communication, demonstrated in Nomadic Radio, provides a simple and constrained problem domain in which to develop and evaluate a contextual notification model. Messaging requires development of a model that dynamically selects a suitable notification strategy based on message priority, usage level, and environmental context. Such a system must infer the user's attention by monitoring her current activities such as interactions with the device and conversations in the room. The user's prior responses to notifications must also be taken into consideration to adapt the notifications over time. In this paper, we will consider techniques for scaleable auditory presentation and an appropriate parameterized

approach towards contextual notification.

Several recent projects utilized speech and audio I/O on wearable devices to present information. A prototype augmented audio tour guide [l] played digital audio recordings indexed by the spatial location of visitors in a museum. SpeechWear [11] enabled users to perform data entry and retrieval using speech recognition and synthesis. Audio Aura [10] explored the use of background auditory cues to provide serendipitous information coupled with people's physical location in the workplace. In Nomadic Radio, the user's inferred context rather than actual location is used to decide when and how to deliver scaleable audio notifications. In a recent paper [13], researchers suggest the use of sensors and user modeling to allow wearables to infer when users should be interrupted by incoming messages. They suggest waiting for a break in the conversation to post a message summary on the user's heads-up display. In this paper we describe a primarily non-visual approach to provide timely information to nomadic listeners, based on a variety of contextual cues.

Nomadic Radio is a wearable computing platform that provides a unified audio-only interface to remote services and messages such as email, voice mail, hourly news broadcasts, and personal calendar events. These messages are automatically downloaded to the device throughout the day and users can browse through them using voice commands and tactile input. The system consists of Java-based clients and remote servers (written in C and Perl) that communicate over wireless LAN, and utilize the telephony infrastructure in the Speech Interface group. Simultaneous spatial audio streams are rendered using a HRTF-based Java audio API. Speech I/O is provided via a networked implementation of AT&T Watson Speech API.

To provide a hands-free and unobtrusive interface to a nomadic user, the system primarily operates as a wearable audio-only device. The SoundBeam Neckset, a research prototype patented by Nortel for use in hands-free telephony, was adapted as the primary wearable platform in Nomadic Radio. It consists of two directional speakers mounted on the user's shoulders, and a directional

microphone placed on the chest (see figure 1). Here information and feedback is provided to the user through a combination of auditory cues, spatial audio rendering, and synthetic speech. Integration of a variety of auditory techniques on a wearable device provides hands-free access and navigation as well as lightweight and expressive notification.

An audio-only interface has been incorporated in Nomadic Radio, and a networked infrastructure for unified messaging has been developed for wearable access [12]. The system currently operates on a Libretto 100 mini-portable PC worn by the user. The key issue addressed in this paper is that of handling interruptions to the listener in a manner that reduces disruption, while providing timely notifications for contextually relevant messages.

USAGE AND NOTIFICATION SCENARIO

The following scenario demonstrates the audio interface and presentation of notifications in Nomadic Radio (no voice commands from the user are shown here).

SCALEABLE AUDITORY PRESENTATION

A scaleable presentation is necessary for delivering sufficient information while minimizing interruption to the listener. Messages in Nomadic Radio are scaled dynamically to unfold as seven increasing levels of notification (see figure 3): silence, ambient cues, auditory cues, message summary, preview, full body, and foreground rendering. These are described further below:

Silence for Least Interruption and Conservation

In this mode all auditory cues and speech feedback are turned-off. Messages can be scaled down to silence when the message priority is inferred to be too low for the message to be relevant for playback or awareness to a user, based on her recent usage of the device and the conversation level. This mode also serves to conserve processing, power and memory resources on a portable device or wearable computer.

Ambient Cues for Peripheral Awareness

In Nomadic Radio, ambient auditory cues are continuously

played in the background to provide an awareness of the operational state of the system and ongoing status of messages being downloaded (see figure 4). The sound of flowing water provides an unobtrusive form of ambient awareness that indicates the system is active (silence indicates sleep mode). Such a sound tends to fade into the perceptual background after a short time, so it does not distract the listener. The pitch is increased during file downloads, momentarily foregrounding the ambient sound. A short e-mail message sounds like a splash while a two-minute audio news summary is heard as faster flowing water while being downloaded. This implicitly indicates message size without the need for additional audio cues and prepares the listener to hear (or deactivate) the message before it becomes available. Such peripheral awareness minimizes cognitive overhead of monitoring incoming messages relative to notifications played as distinct auditory cues, which incur a somewhat higher cost of attention on part of the listener.

Related Work in Auditory Awareness

In ARKola [5], an audio/visual simulation of a bottling factory, repetitive streams of sounds allowed people to keep track of activity, rate, and functioning of running machines. Without sounds people often overlooked problems; with auditory cues, problems were indicated by the machine's sound ceasing (often ineffective) or via distinct alert sounds. The various auditory cues (as many as 12 sounds play simultaneously) merged as an auditory texture, allowed people to hear the plant as a complex integrated process.

Background sounds were also explored in ShareMon [3], a prototype application that notified users of file sharing activity. Cohen found that pink noise used to indicate %CPU time was considered "obnoxious", even though users understood the, pitch correlation. However, preliminary reactions to wave sounds were considered positive and even soothing. In Audio Aura [IO], alarm

sounds were eliminated and a number of "harmonically coherent sonic ecologies" were explored, mapping events to auditory, musical or voice-based feedback. Such techniques

were used to passively convey the number of email messages received, identity of senders, and abstract representations of group activity.

Auditory Cues for Notification and Identification

In Nomadic Radio, auditory cues are a crucial means for conveying awareness, notification and providing necessary assurances in its non-visual interface. Different types of auditory techniques provide distinct feedback, awareness and message information.

Feedback Cues

Several types of audio cues indicate feedback for a number of operational events in Nomadic Radio:

1. Task completion and confirmations - button pressed, speech understood, connected to servers, finished playing or loaded/deleted messages.
2. Mode transitions - switching categories, going to non-speech or ambient mode.
3. Exceptional conditions - message not found, lost connection with servers, and errors.

Priority Cues for Notification

In a related project, “email glances” [7] were formulated as a stream of short sounds indicating category, sender and content flags (from keywords in the message). In Nomadic Radio, message priority inferred from email content filtering provides distinct auditory cues (assigned by the user) for group, personal, timely, and important messages. In addition, auditory cues such as telephone ringing indicate voice mail, whereas an extracted sound of a station identifier indicates a news summary.

VoiceCues for Identification

VoiceCues represent a novel approach for easy identification of the sender of an email, based on a unique auditory signature of the person. VoiceCues are created by manually extracting a l-2 second audio sample from the voice messages of callers and associating them with their respective email login. When a new email message arrives, the system queries its database for a related VoiceCue for that person before playing it to the user as a notification, along with the priority cues. The authors have found VoiceCues to be a remarkably effective method for quickly

conveying the sender of the message in a very short duration. This technique reduces the need for synthetic speech feedback, which can often be distracting.

Message Summary Generation

A spoken description of an incoming message can present relevant information in a concise manner. Such a description typically utilizes header information in email messages to convey the name of the sender and the subject of the message. In Nomadic Radio, message summaries are generated for all messages, including voice-mail, news and calendar events. The summaries are augmented by additional attributes of the message indicating category, order, priority, and duration. For audio sources, like voice messages and news broadcasts, the system plays the first 2.5 seconds of the audio. This identifies the caller and the urgency of the call, inferred from intonation in the caller's voice or provides a station identifier for news summaries.

Message Previews using Content Summarization

Messages are scaled to allow listeners to quickly preview the contents of an email or voice message. In Nomadic Radio, a preview for text messages extracts the first 100 characters of the message (a default size that can be user defined). This heuristic generally provides sufficient context for the listener to anticipate the overall message theme and urgency. For email messages, redundant headers and previous replies are eliminated from the preview for effective extraction. Use of text summarization techniques, based on tools such as ProSum' developed by British Telecom, would allow more flexible means of scaling message content. Natural language parsing techniques used in ProSum permit a scaleable summary of an arbitrarily large text document.

A preview for an audio source such as a voice message or news broadcast presents a fifth of the message at a gradually increasing playback rate of up to 1.3 times faster than normal. There are a range of techniques for time-compressing speech without modifying the pitch, however twice the playback rate usually makes the audio

incomprehensible. A better representation for content summarization requires a structural description of the audio, based on annotated or automatically determined pauses in speech, speaker and topic changes. Such an auditory thumbnail must function similar to its visual counterpart. A preview for a structured voice message would provide pertinent aspects such as name of caller and phone number, whereas a structured news preview would be heard as the hourly headlines.

Full Body: Playing Complete Message Content

This mode plays the entire audio file or reads the full text of the message at the original playback rate. Some parsing of the text is necessary to eliminate redundant header information and format tags. The message is augmented with summary information indicating sender and subject. This message is generally spoken or played in the background of the listener's audio space.

I http://transend.Iabs.bt.com/prosum/on-line/

Foreground Rendering via Spatial Proximity

An important message is played in the foreground of the listening space. The audio source of the message is rapidly moved closer to the listener, allowing it to be heard louder, and played there for 415" of its duration. The message gradually begins to fade away, moving back to its original position and amplitude for the remaining l/S" of the duration. The foregrounding algorithm ensures that the messages are quickly brought into perceptual focus by pulling them to the listener rapidly. However the messages are pushed back slowly to provide an easy fading effect as the next one is heard. As the message moves its spatial direction is maintained so that the listener can retain a focus on the audio source even if another begins to play.

Hence a range of techniques provide scaleable forms of background awareness, auditory notification, spoken feedback and foreground rendering of incoming messages.

CONTEXTUAL NOTIFICATION

In Nomadic Radio, context dynamically scales the notifications for incoming messages. The primary contextual cues used include: message priority from email filtering, usage level based on time since last user action,

and the likelihood of conversation estimated from real-time analysis of the auditory scene. In our experience these parameters provide sufficient context to scale notifications, however data from motion or location sensors can also be integrated in such a model. A linear and scaleable auditory notification model is utilized, based on the notion of estimating costs of interruption and the value of information to be delivered to the user. This approach is similar to recent work [6] on using perceptual costs and a focus of attention model for scaleable graphics rendering.

Message Priority

The priority of incoming messages is explicitly determined via content-based email filtering using CLUES [9], a filtering and prioritization system. CLUES has been integrated into Nomadic Radio to determine the timely nature of messages by finding correlation between a user's calendar, rolodex, to-do list, as well as a record of outgoing messages and phone calls. These rules are integrated with static rules created by the user for prioritizing specific people or message subjects. When a new email message arrives, keywords from its sender and. subject header information are correlated with static and generated filtering rules to assign a priority to the message. Email messages are also prioritized if the user is traveling and meeting others in the same geographic area (via area codes in the rolodex). The current priorities include: group, personal, very important, most important, and timely. Priorities are parameterized by logarithmically scaling all priorities within a range of 0 to 1. Logarithmic scaling ensures that higher priority messages are weighted higher relative to unimportant or uncategorized messages.

Priority (i) = (log (i) / log (Priority Levels Mu))

Usage Level

One problem with using last actions for setting usage levels is that if a user deactivates an annoying message, that action is again time-stamped. Such negative reinforcements continue to increase the usage level and the related notification. Therefore negative actions such as stopping

audio playback or deactivating speech are excluded from generating actions for computing the usage.

Likelihood of Conversation

Conversation in the environment can be used to gauge whether the user is in a social context where an interruption is less appropriate. If the system detects the occurrence of more than several speakers over a period of time, that is an indication of a conversational situation. Auditory events are first detected by adaptively thresholding total energy and incorporating constraints on event length and surrounding pauses. The system uses mel-scaled filter-bank coefficients (MFCs) and pitch estimates to discriminate, reasonably well, a variety of speech and non-speech sounds. HMMs (Hidden Markov Models) capture both the temporal characteristics and spectral content of sound events. The techniques for feature extraction and classification of the auditory scene using HMMs are described in a recent workshop paper [2]. The likelihood of speech detected in the environment is computed for each event in a short window of time. In addition, the probabilities are weighted, such that most recent time periods in the window are considered more relevant for computing the overall Speech Level. We are evaluating the classifier's effectiveness by training it with a variety of speakers and background sounds.

Notification Level

A weighted average for all three contextual cues provides level has an inversely proportional relationship with notification i.e. a lower notification must be provided during high conversation.

Presentation Latency

Latency represents the period of time to wait before playing the message to the listener, after a notification cue is delivered. Latency is computed as a function of the notification level and the maximum window of time (Latency,& that a lowest priority message can be delayed for playback. The default maximum latency is set to 20 seconds, but can be modified by the user.

were increased. Jane was notified of a group message shortly after the voice message, since the system detected higher usage activity. Hence, the system correctly scaled down notifications when Jane did not want to be bothered whereas notifications were scaled up when Jane started to use the system to browse her messages.

EFFECTIVENESS OF THE NOTIFICATION MODEL

The nature of peripheral awareness and unobtrusive notification on a wearable device requires a usage evaluation that must be conducted on an ongoing and long-term basis. However, the predictive effectiveness of the notification model must first be evaluated on a quantitative basis. Hence, all message and notification parameters are captured for such analysis. Lets consider two actual examples of notification computed for email messages with different priorities. Figure 7 shows an auditory cue generated for a group message (low priority).

The timely message (in figure 8) received greater priority and consequently a higher notification level for summary playback. A moderate latency time (approx. 6 secs.) was chosen. However when the user interrupted the notification by a button press, the summary playback was aborted. The user's action reduced overall weights by 5%.

Dynamic Adaptation of the Notification Model

The user can initially set the weights for the notification model to high, medium, or low (interruption). These weight settings were selected by experimenting with notifications over time using an interactive visualization of message parameters. This allowed us to observe the model, modify weights and infer the effect on notification based on different weighting strategies. Pre-defined weights provide an approximate behavior for the model and help bootstrap the system for novice users. The system also allows the user to dynamically adjust these weights (changing the interruption and notification levels) by their implicit actions while playing or ignoring messages.

The system allows localized positive and negative reinforcement of the weights by monitoring the actions of the user during notifications. As a message arrives, the

system plays an auditory cue if its computed notification level is above the necessary threshold for auditory cues. It then uses the computed latency interval to wait before playing the appropriate summary or preview of the message. During that time, the user can request the message be played earlier or abort any further notification for the message via speech or button commands. If aborted, all weights are reduced by a fixed percentage (default is 5%), a negative reinforcement. If the user activates the message (positive reinforcement) within 60 seconds after the notification, the playback scale selected by the user is used to increase all weights. If the message is ignored, no change is made to the weights, but the message remains active for 60 seconds during which the user's actions can continue to influence the weights.

Figure 6 shows a zoomed view of the extended scenario introduced earlier, focusing on Jane's actions that reinforce the model. Jane received several messages and ignored most of the group messages and a recent personal message (the weights remain unchanged). While in the meeting, Jane interrupted a timely message to abort its playback. This reduced the weights for future messages, and the ones with low priority (group message) were not notified to Jane. The voice message from Kathy, her daughter, prompted Jane to reinforce the message by playing it. In this case, the weights Continuous local reinforcement over time should allow the system to reach a state where it is somewhat stable and robust in converging to the user's preferred notification. Currently the user's actions primarily adjust weights for subsequent messages, however effective reinforcement learning requires a model that generalizes a notification policy that maximizes some long-term measure of reinforcement [8]; this will be the focus of our future work.

PRELIMINARY EVALUATION

Although the authors have been using and relining these techniques during system development, a preliminary 2-day evaluation was conducted with a novice user, who had prior experience with mobile phones and 2-way pagers. The user was able to listen to notifications while attending to tasks in

parallel such as reading or typing. He managed to have casual discussions with others while hearing notifications; however he preferred turning off all audio during an important meeting with his advisor. People nearby sometimes found the spoken feedback distracting if heard louder, however that also cued them to wait before interrupting the user. The volume on the device was lowered to minimize any disruption to others and maintain the privacy of messages. The user requested an automatic volume gain that adapted to the environmental noise level.

In contrast to speech-only feedback, the user found the unfolding presentation of ambient and auditory cues allowed sufficient time to switch attention to the incoming message. Familiarization with the auditory cues was necessary. He preferred longer and gradual notifications rather than distinct auditory tones. The priority cues were the least useful indicator whereas VoiceCues provided obvious benefit. Knowing the actual priority of a message was less important than simply having it presented in the right manner. The user suggested weaving message priority into the ambient audio (as increased pitch). He found the overall auditory scheme somewhat complex, preferring instead a simple notification consisting of ambient awareness, Voice&es and spoken text.

The user stressed that the ambient audio provided the most benefit while requiring least cognitive effort. He wished to hear ambient audio at all times to remain reassured that the system was still operational. An unintended effect discovered was that a “pulsating” audio stream indicated low battery power on the wearable device. A “pause” button was requested, to hold all messages while participating in a conversation, along with subtle but periodic auditory alerts for unread messages waiting in queue. The user felt that Nomadic Radio provided appropriate awareness and its expressive qualities justified its use over a pager. A long-term trial with several nomadic users is necessary to further validate these notification techniques.

CONCLUSIONS

We have demonstrated techniques for scaleable auditory presentation and message notification using a variety of

contextual cues. The auditory techniques and notification model have been refined based on continuous usage by the authors, however we are currently conducting additional evaluations with several users. Ongoing work explores adaptation of the notification model based on reinforcement from user behavior over time. Our efforts have focused on wearable audio platforms, however these ideas can be readily utilized in consumer devices such as pagers, PDAs and mobile phones to minimize disruptions while providing timely information to users on the move.

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