

UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

SAMSUNG ELECTRONICS CO., LTD., SAMSUNG ELECTRONICS,
AMERICA, INC., HARMAN INTERNATIONAL INDUSTRIES, INC.,

Petitioners

v.

STATON TECHIYA, LLC,

Patent Owner

IPR2024-00559
U.S. Patent No. 11,610,587

DECLARATION OF DR. ERIC TARR on behalf of PATENT OWNER

Mail Stop “PATENT BOARD”
Patent Trial and Appeal Board
U.S. Patent and Trademark Office
P.O. Box 1450
Alexandria, VA 22313-1450

TABLE OF CONTENTS

I.	INTRODUCTION	3
II.	QUALIFICATIONS	4
III.	MATERIALS CONSIDERED	6
IV.	LEGAL PRINCIPLES	6
V.	THE LEVEL OF ORDINARY SKILL IN THE ART	7
VI.	BACKGROUND	8
	A. The ‘587 Patent.....	8
	B. U.S. Pat. No. 6,804,638 (“Fiedler”) (EX1008).....	11
	C. U.S. Pub. No. 2006/0195322 (“Broussard”) (EX1008).....	14
	D. European Pat. Appl. No. EP 1 519 625 (“Victorian”) (EX1009)	14
	E. U.S. Pub. No. 2007/0281744 (“Andreasson”) (EX1010)	15
	F. U.S. Pub. No. 2008/0162133 (“Couper”) (EX1011)	16
	G. U.S. Pub. No. 2003/0032447 (“Bulthuis”) (EX1012)	16
VII.	SAMSUNG’s Petition Fails to Establish that Claims 1-12 are UNPATENTABLE	17
	A. Ground 1A: Fiedler and/or Broussard does not render Claims 1, 3, 4 or 12 of the ‘587 Patent Unpatentable.	17
	1. Fiedler Does Not Teach “Sending A Portion of the Microphone Signal to the Audio Buffer”	17
	2. Fiedler Does Not Disclose “Sending A Modified Microphone Signal To The Data Buffer...”	24
	3. Claims 3, 4 and 12	25
	B. Ground 1B: Fiedler, Broussard and Victorian Do Not Render Claims 5-9 of the ‘587 Patent Obvious.....	26
	C. Ground 1C: Fiedler, Broussard, Victorian and Andreasson Do Not Render Claim 8 of the ‘587 Patent Obvious	26
	D. Ground 1D: Fiedler, Broussard and Couper Do Not Render Claims 10 and 11 of the ‘587 Patent Obvious	27
	E. Ground 2: Fiedler, Broussard and Bulthuis Do Not Render Claims 1 and 2 of the ‘587 Patent Obvious	28

I. INTRODUCTION

1. My name is Dr. Eric Tarr. I have been retained by Staton Techiya LLC, which I may refer to as either the “Patent Owner” or “Techiya,” for this inter partes review proceeding, which I may refer to as “this IPR” or “the IPR.” I understand that this IPR involves U.S. Patent No. 11,610,587 (EX1001), which I may refer to as “the ‘587 patent” for shorthand. I understand that the claims challenged in this IPR are claims 1-12 of the ‘587 patent.

2. I understand that the ‘587 patent is assigned to Techiya.

3. I understand that in this proceeding Samsung Electronics Co., Ltd., Samsung Electronics America, Inc., and Harman International Industries, Inc., which I may also refer to collectively as the “Petitioner” or “Samsung,” filed a Petition for review of Claims 1-12 of the ‘587 patent.

4. I understand that the above claims are challenged by Petitioner based upon anticipation and/or obviousness over the following references:

Claim(s) Challenged	35 U.S.C. §	Reference(s)/Basis
1, 3, 4, 12	103(a)	Fiedler, Broussard
5-9	103(a)	Fiedler, Broussard, Victorian
8	103(a)	Fiedler, Broussard, Victorian, Andreasson
10-11	103(a)	Fiedler, Broussard, Couper
1-2	103(a)	Fiedler, Broussard, Bulthuis

5. I have been asked to provide my objective, independent analysis of the ‘587 patent in view of the asserted prior art references and other documents cited in the Petition and to provide my opinion regarding the allegations in the Petition, as well as the supporting opinions of Christopher Schmandt (EX1002).

6. I am not currently, nor have I ever been, an employee of Techiya. I received no compensation for this declaration beyond my normal hourly compensation of \$300 based on my time actually spent analyzing the ‘587 patent, the asserted prior art references cited herein and the Petition, and any related issues. I will not receive any added compensation based on the outcome of this IPR, any other IPR, or any other proceeding involving the ‘587 patent.

7. Based upon the materials that I have considered, and the arguments made by Petitioner and Mr. Schmandt, as explained below it is my opinion that Petitioner has failed to meet its burden to show that any challenged claim of the ’587 Patent is unpatentable.

II. QUALIFICATIONS

8. My general qualifications can be found in my curriculum vitae attached as Appendix A. It includes my educational and professional background, published books, articles, publications, public lectures and conference proceedings. One section details my assignments testifying at depositions and trial as an expert witness.

9. I am an associate professor of Audio Engineering Technology at Belmont University in Nashville, Tennessee. I received a B.S., M.S., and Ph.D. in Electrical and Computer Engineering from the Ohio State University with a specialization in digital signal processing.

10. I am the author of a textbook published by the international professional society of audio engineers titled, “Hack Audio: An Introduction to Computer Programming and Digital Signal Processing in MATLAB.”

11. I regularly teach classes involving audio software development, acoustics, as well as the anatomy and physiology of the auditory system. I have consulted for several companies developing commercial audio software products.

12. I have no financial interest in either party or in the outcome of this proceeding. I am being paid for my work as an expert in these matters on an hourly basis. My compensation is not dependent on the outcome of these proceedings or the content of my opinions. My opinions, as explained below, are based on my education, experience, and background in the fields discussed above, and my review of the '587 Patent and its file history, the materials provided by Patent Owner, and the other materials discussed herein.

13. For consistency and ease of review, all of my column and line citations to the patent specifications are in “column:line” format.

III. MATERIALS CONSIDERED

14. I have reviewed the '587 Patent, including the challenged claims, and its prosecution history. I have also reviewed the Petition for inter partes review filed by Petitioner, as well as EX1001-1035 attached.

15. In forming my opinions, I have considered the materials listed above and any documents cited in this declaration. I have also relied on my own education, knowledge, and experience in the relevant art. I have also considered the understanding of a POSITA around the time of the invention of the '587 patent.

IV. LEGAL PRINCIPLES

16. I have been instructed that claims in an inter partes review are construed using the same claim construction standard as in a civil action, and that under this standard, claims terms are generally given their ordinary and customary meaning as understood by a person of ordinary skill in the art ("POSITA") at the time of the invention.

17. The POSITA is deemed to read the claim term in the context of the entire patent, including the specification and prosecution history.

18. I have been informed that a POSITA departs from the plain and ordinary meaning in only two instances: (1) when a patentee acts as its own lexicographer, and (2) when the patentee disavows the full scope of the claim term in the specification or during prosecution.

19. I have been informed that in order for a claim to be rendered obvious by a single reference, or a combination of two or more references, the following factors must be considered: 1) the scope and content of the prior art; 2) the differences between the prior art and the claims at issue; 3) the level of ordinary skill in the art; and 4) any secondary considerations of non-obviousness. An obviousness determination cannot be sustained with mere conclusory statements; instead, there must be some articulated reasoning with some rational underpinning to support the legal conclusion of obviousness.

V. THE LEVEL OF ORDINARY SKILL IN THE ART

20. In my opinion, a person of ordinary skill in the art (“POSITA”) would have had a minimum of a bachelor’s degree in electrical, computer science, or a related field, along with at least two years of professional experience in the design of digital audio systems and associated signal processing such as voice or speech processing and/or acoustic pattern recognition (or additional graduate education substituting for professional experience and/or significant experience in the field substituting for formal education).

21. I am a person of ordinary skill in the art. As explained above, I have a Ph.D. in electrical engineering and over 10 years in the design digital audio systems and associated signal processing such as voice or speech processing and/or acoustic pattern recognition.

VI. BACKGROUND

A. The '587 Patent

22. The '587 patent is entitled “Personalized Sound Management System and Method.” The specification of the '587 patent states that it “relates in general to methods and systems for implementing a suite of personalized sound applications for modifying a user’s acoustic environment...and providing user selection and control over the management of sound.” 1:43-50. The patent teaches systems for what the '587 patent refers to as “personalized sound management” in which an individual can “modify what is received from the acoustic space [] thereby tailoring or personalizing the received acoustic information to meet the needs of [the] person [].” 3:44-47. The individual “selects the applications and control devices [] to modify acoustic space [] to meet their personal needs and wants.” 3:53-55.

23. The '587 patent explains that “[i]n general, humans are continuously inundated with a diversity of sounds. Many of the sounds are not critical to our lives but our brain processes these sounds and tries to distinguish between them.” 1:33-36. This diversity of sound “can cause an auditory overload that can impact both the health and safety of an individual.” EX1001, 1:38-39; *see also* 3:62-67.

24. To protect against this, the inventions of the '587 patent allow the user to “modify a user’s acoustic environment” and provid[e] user selection and control over the management of sound.” *Id.* at 1:45, 49-50.

25. The patent discloses a personalized sound management system 206, for which several embodiments are disclosed. Systems implementing these applications may each include a microphone configured to measure ambient sound and generate a microphone signal, a speaker configured to emit an audio signal, a memory configured to store instructions, an audio buffer configured to store at least a portion of the microphone signal, a data buffer; and a processor operatively coupled to the memory, the audio buffer, and the data buffer. The processor is configured to execute the instructions to receive the microphone signal, send a portion of the microphone signal to the audio buffer, analyze the audio buffer in order to detect a sonic signature, notify a user when a sonic signature is detected, send a modified microphone signal to the data buffer that replaces or adds to previous data stored in the data buffer, and sends a portion of the data in the buffer, spanning a time period, to the speaker in response to a user request.

26. In one embodiment, the Safe Space system modifies the user's acoustic environment by listening for specific sounds (e.g., the siren of an emergency vehicle) in the external acoustic environment and providing a response in the internal acoustic environment. EX1001, 5:19-38.

27. In another embodiment, the Earguard system, a system is disclosed in which the external acoustic environment is monitored using a sound pressure level (SPL) dose management system, and allows for modification of the internal acoustic

environment of the user either automatically or manually when the SPL dose reaches a level that could damage the ear. EX1001, 5:59-6:5.

28. In another embodiment, a signal containing the user's voice is modified so as to be more natural sounding and easily recognizable to another hearing the voice. Similarly, the user can modify their acoustic environment to be able to clearly hear both their own voice and the voices of others on, e.g., a telephone call by eliminating or minimizing background noise in a noisy external environment.*Id.*, 6:47-7:1, 7:8-17.

29. As another example, in the Earbliss system, a personalized sound management application is provided to acoustically isolate someone who snores from another's acoustic environment, while allowing sounds other than snoring to be heard normally. *Id.*, 7:48-51, 8:16-31.

30. In yet another embodiment, the Hear and Tell system, among the operations the device can perform are "passing the detected signal through to the user, boosting the sound such that the user is made aware of the detected sound, rejecting the sound so that the user does not hear the detected sound, attenuating the detected sound, and replacing the detected sound with an alternate sound to name. *Id.*, 9:21-27.

B. U.S. Pat. No. 6,804,638 (“Fiedler”) (EX1008)

31. U.S. Patent No. 6,804,638 to Fiedler is entitled “Device and Method For Selective Recall and Preservation of Events Prior To Decision To Record The Events.” Fiedler discloses that its invention “enables users effectively to record words that have already been spoken...by the user.” 2:35-44. As disclosed in FIG. 1, “a microphone feed(s)...circular set of acquisition buffers.” 3:67-4:4. The microphone signal is then sent from the input buffer to a “first acquisition buffer.” See FIG. 2; 5:10-17; 5:24-36.

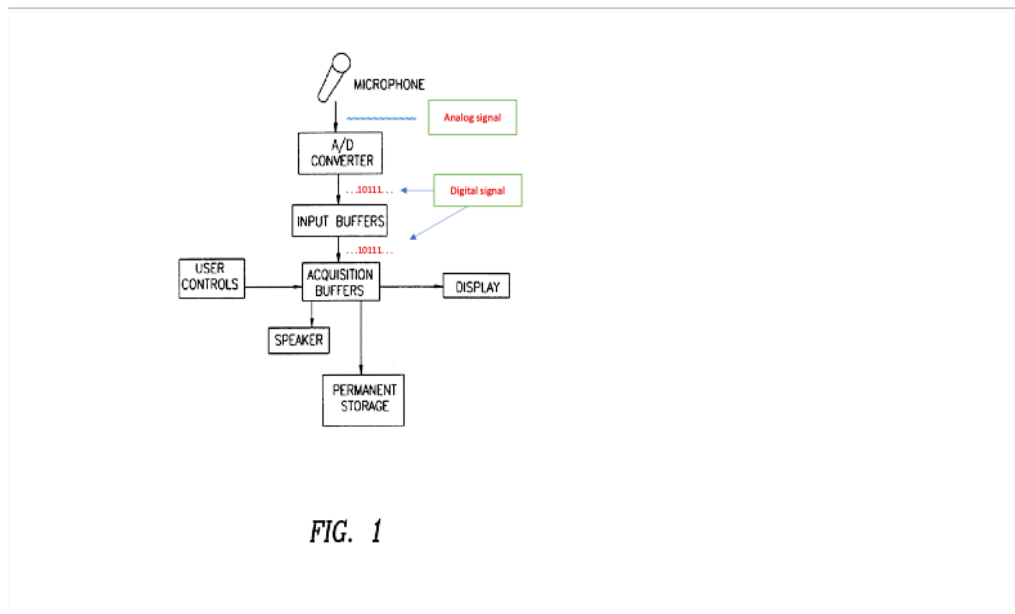
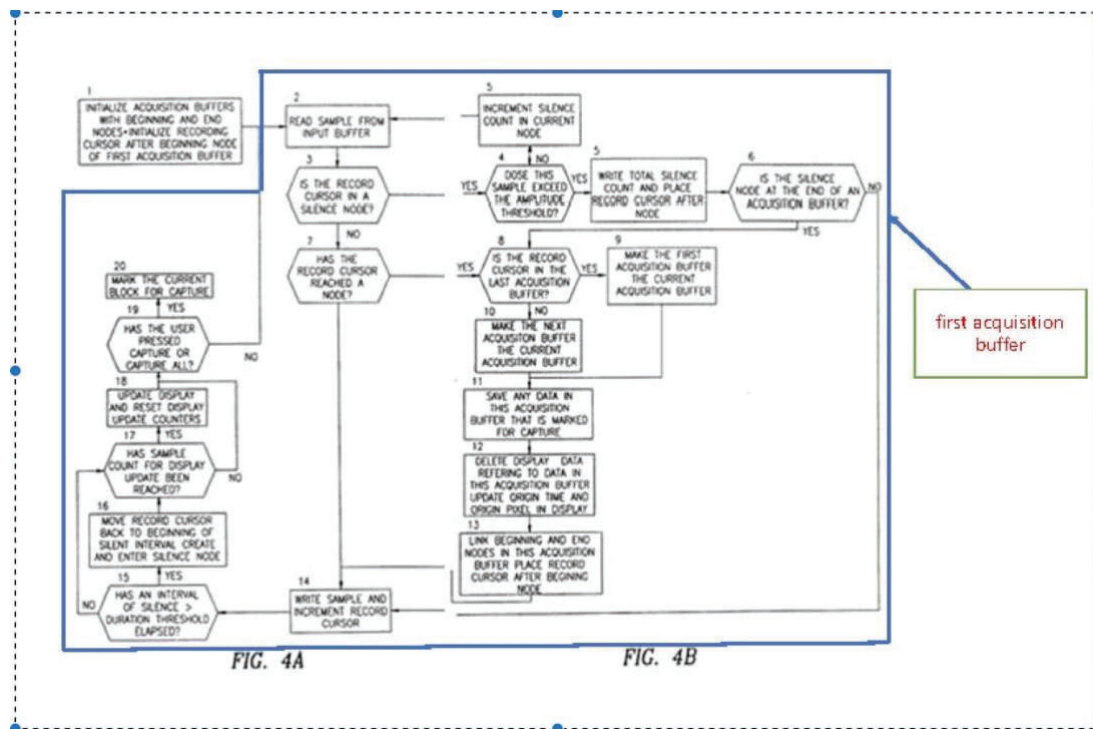


FIG. 4 of Fiedler illustrates the process by which the microphone signal is transmitted from the input buffer to the first acquisition buffer.

32. Fiedler discloses that “[a]s audio data are transferred from the input buffers to the acquisition buffers, the successive amplitude values in the audio stream are measured against the amplitude and duration thresholds (FIG. 4, block 15), and the stream is accordingly stored in the acquisition buffers as segments of sound data interspersed with silence nodes.” “[A]fter each sample of audio data is initially written to the acquisition buffer...the recording cursor goes back to the beginning of the sequence of amplitude values in the acquisition buffers corresponding to this interval and writes a silence node there.” EX1007, 5:54-59, 6:1-10.



33. Fielder proposes, in addition to the primary embodiment above, a second alternative embodiment which can be used for “event-activated recording,”

which is effected “on detection of a suitable triggering event.” 6:33-35. In this alternative embodiment, “the portion of the acquisition buffer contents leading up to the triggering even may be transferred to a second acquisition buffer, which will be used in place of the original acquisition buffer for all purposes described below, including the presentation of data to the user for selection for recall and/or preservation.” Petition at 34; EX1002 at 59 (citing EX1007, 6:42-47).

34. In this alternative embodiment, the information in the first acquisition buffer is transferred to a second acquisition buffer from a time “just prior to the triggering event, and advantageously forms a continuous record with the triggering event and the events following it.” Pet. at 35 (citing EX1002 at 59 (further citing EX1007, 6:50-52)).

35. Fiedler discloses that the process by which this record is created is effected by: (1) continuing to record in the acquisition buffer the triggering event and the events following the triggering event, (2) concurrently with this continuous recording, copying the appropriate portions of the acquisition buffer contents to [the second acquisition buffer], preserving the record of events prior to, including, and following the triggering event, and (3) recording events directly to [the] second acquisition buffer] when the copying process has caught up with the direct recording process by reaching the present moment.” *Id.*, 6:56-67.

36. It is my understanding that Petitioner and Mr. Schmandt acknowledge that their purported combination of Fiedler with the remaining secondary references is based on their specific, alternative configuration of Fiedler. Pet. at 16, 34, 36-38. According to the Petition, the first acquisition buffer of Fiedler corresponds to the “audio buffer” of the ‘587 patent, while the “second acquisition buffer” of the alternative embodiment corresponds to the “data buffer” of the ‘587 patent. *Id.*

C. U.S. Pub. No. 2006/0195322 (“Broussard”) (EX1008)

37. United States Patent Application Pub. No. 2006/0195322 to Broussard (EX1008) is entitled “System and Method for Detecting and Storing Important Information.” The publication discloses “an improved method for recording audio notes for easier later retrieval.” EX1008, Abstract. “FIG. 2 is a block diagram of major components of the processing and storage unit 12.” *Id.*, [0015]. The unit system would either have to receive the analog signal in digital form or have an analog to Digital (A to D) converter 32 which makes the data available to a data processor 34...”. *Id.* “After having been converted to a digital audio signal as previously described, the audio signal 60 enters a buffer memory 62. A trigger detection subsystems 64 uses the data in the buffer 62 to look for triggers in the data that indicate that the incoming signal contains information which should be recorded in a separate extended audio segment.” *Id.*, [0019].

D. European Pat. Appl. No. EP 1 519 625 (“Victorian”) (EX1009)

38. European Patent Application No. EP 1519625 A2 to Victorian et al. (EX1009) is entitled, “External ear canal voice detection,” and discloses an “ear-level full duplex audio communication systems” which “include one or two ear attachment devices...that wirelessly communicates to a remote device such as a computer, a personal digital assistant (PDA), a cellular phone, a walkie talkie or a language translator.” EX1009, Abstract.

39. Victorian discloses that “[i]n one embodiment of the invention, a system includes an earmold configured for use as an [in-the-ear] device,” which includes a microphone that receives an “occluded sound” (i.e., the user’s voice) from about the ear canal, processes the sound, and transmits it, while receiving an incoming wireless signal representing a remote sound and converting it into the remote sound. *Id.*, [0009]-[0011].

40. Victorian does not disclose receipt of any wireless signal other than one representing a remote sound.

E. U.S. Pub. No. 2007/0281744 (“Andreasson”) (EX1010)

41. U.S. Application No. 2007/0281744 A1 to Andreasson (EX1010) is titled, “Audio Output Device Selection For A Portable Electronic Device,” and discloses “detect[ing] whether an output unit of an accessory is connected to the body of a user and determines audio output units to which its application will transmit audio signal [sic].” EX1010, Abstract.

42. Andreasson discloses embodiments including an in-the-ear and/or behind-the-ear module.

F. U.S. Pub. No. 2008/0162133 (“Couper”) (EX1011)

43. U.S. Application No. 2008/0162133 A1 to Couper et al. (EX1011) is entitled “Audio Detection Using Distributed Mobile Computing,” and discloses “a method of identifying incidents using mobile devices.” EX1011, [0004]. Such incidents are identified by sound signatures 120, such as “gunshots, explosions, sirens...” *Id.*, [0026].

44. Upon determining that a sound 150 received by a mobile matches a sound signature 120, “the mobile device [] can send a communication to the event processor 125. In one embodiment, the communication can be a Short Message Service (SMS) message 155.” *Id.*, [0034]. The mobile device can also send a recording of the detected sound over the voice channel of the network. *Id.*, [0044].

G. U.S. Pub. No. 2003/0032447 (“Bulthuis”) (EX1012)

45. U.S. Application No. 2003/0032447 A1 to Bulthuis (EX1012) is entitled “Conversation Rewind,” and discloses “a consumer electronics device...designed to record all real time communications occurring on it.” EX1012, Abstract. “Recording is started when a communication is initiated, either by the user making or accepting a call.” *Id.*

46. Petitioner does not allege that Bulthuis discloses the use of any form of an audio buffer or data buffer.

VII. SAMSUNG’S PETITION FAILS TO ESTABLISH THAT CLAIMS 1-12 ARE UNPATENTABLE

47. Samsung’s Petition includes five grounds as shown below.

Ground	References	Claims
1A	Fiedler, Broussard	1, 3, 4, 12
1B	Fiedler, Broussard, Victorian	5-9
1C	Fiedler, Broussard, Victorian, Andreasson	8
1D	Fiedler, Broussard, Couper	10-11
2	Fiedler, Broussard, Bulthuis	1-2

48. It is my opinion that the Petition fails to show that any of the Grounds set forth by Petitioner render any of claims 1-12 of the ‘587 patent unpatentable.

A. Ground 1A: Fiedler and/or Broussard does not render Claims 1, 3, 4 or 12 of the ‘587 Patent Unpatentable.

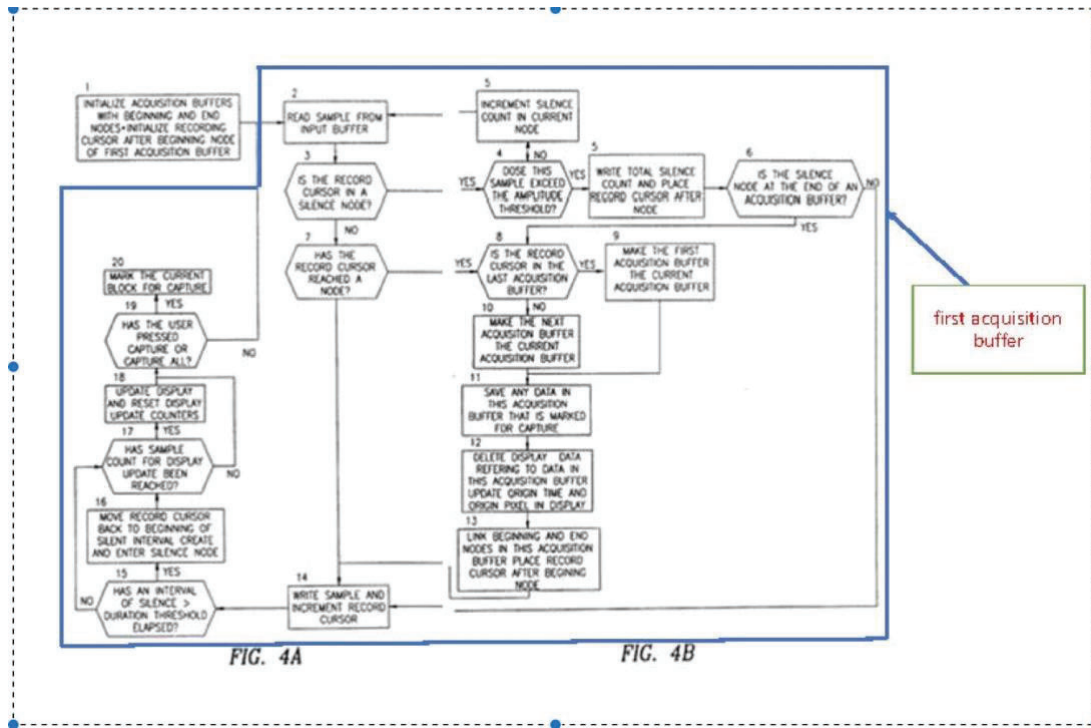
1. Fiedler Does Not Teach “Sending A Portion of the Microphone Signal to the Audio Buffer”

49. It is my understanding that Petitioner and Mr. Schmandt rely on the Fiedler reference to show that the limitation “sending a portion of the microphone signal to the audio buffer” is shown in the prior art.

50. In his declaration, Mr. Schmandt corresponds the first acquisition buffer of Fiedler to the “audio buffer configured to store at least a portion of the microphone signal” in limitation [1.6] of the ‘587 patent. Similarly, Mr. Schmandt

argues that the digital samples received by the input buffer from the analog-to-digital converter are the “microphone signal” of the ‘587 patent, and that each of the digital samples is the ‘587 patent’s claimed “a portion of the microphone signal.” Schmandt para. 160. From this, Petitioner and Mr. Schmandt argue that “transfer of the samples into the original acquisition buffer” meets the limitation “sending a portion of the microphone signal to the audio buffer.” Pet. at 28 (citing EX1002, ¶ 161).

51. In my opinion, this is inconsistent with the disclosures of Fiedler and the ‘587 patent. Fiedler is clear that in both its primary and its alternative embodiment, every “sample” received by Fiedler’s input buffer (i.e., the entire microphone signal according to Petitioner) is sent to Fiedler’s “first acquisition buffer.” See FIG. 2; 5:10-17; 5:24-36. FIG. 4 (reproduced below) illustrates the process:



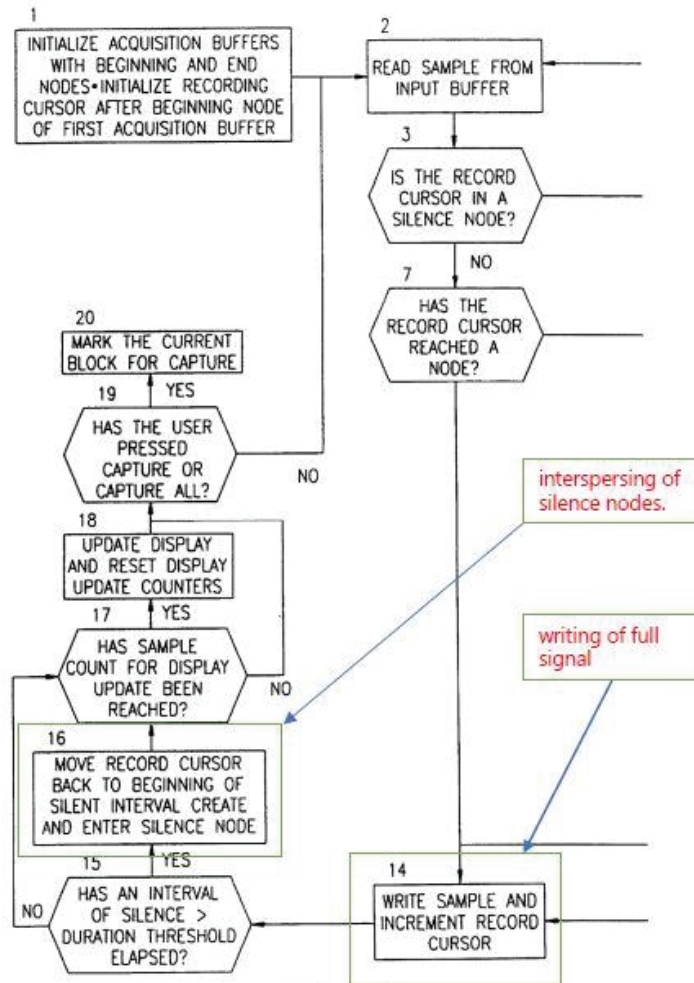


FIG. 4A

EX1001, FIGS. 4A-4B, 3:52-53; *see also* 5:54-59 (“As audio data are transferred from the input buffers to the acquisition buffers, the successive amplitude values in the audio stream are measured against the amplitude and duration thresholds (FIG. 4, block 15), and the stream is accordingly stored in the acquisition buffers as segments of sound data interspersed with silence nodes.)”

52. Claim 1 requires that the “audio buffer” (which Petitioner maps to the first acquisition buffer of Fiedler) be “configured to store at least a portion of the

microphone signal.” EX1001, 24:55-56. In other words, the acquisition buffer must be configured to store more than zero of the microphone signal, but could store the entirety of the microphone signal. In contrast, the claims require that only “a portion of the microphone signal” is actually sent to the audio buffer.

53. One of ordinary skill in the art would understand that the ‘587 patent’s distinguishing between the “at least a portion of the microphone signal” which can be stored in the audio buffer and the “portion of the microphone signal” which is actually sent to the audio buffer to mean that not all of the microphone signal is intended to be sent to the audio buffer in the invention of the ‘587 patent. Because all of the signal received by Fiedler’s input buffer is sent to Fiedler’s acquisition buffer, it is my opinion that Fiedler does not disclose sending “a portion of the microphone signal to the audio buffer” as required by claim 1 of the ‘587 patent.

54. Should Petitioner argue that the “signal” interspersed with silence nodes that is written to the first acquisition buffer of Fielder is the “portion of the microphone signal” in limitation [1.10], I note that Petitioner and Mr. Schmandt later identify the identical signal as the “modified microphone signal” that is sent to the data buffer in limitation [1.13]. Pet. at 36 (“Fiedler’s silence node scheme ‘modifies’ the captured data (i.e., the microphone signal) because it replaces parts of the captured data with a silence node.”) (citing EX1002, ¶ 192).

55. As explained above, it is my understanding that Petitioner and Mr. Schmandt rely on Fiedler's second, alternative embodiment for its argument that the claims of the '587 patent are unpatentable. In that alternative embodiment, Fiedler is clear that after a triggering event, the information that is written to the second acquisition buffer (which Petitioner claims corresponds to the '587 patent's data buffer) is identical to the information contained in the first acquisition buffer (which Petitioner claims corresponds to the audio buffer). 6:48-63. In other words, Petitioner would be taking the position that the same signal is both "a portion of the microphone signal" and "a modified microphone signal."

56. However, the specification of the '587 patent makes clear that the "portion of the microphone signal" and the "modified microphone signal" are two separate signals, used for two separate purposes. While the portion of the microphone signal is used for sonic signature detection, consistent with the language of the claims, the modified microphone signal is used to manage and/or personalize the user's acoustic environment.

57. For example, in one embodiment of the '587 patent, the Safe Space system, the signal that is examined for sonic signature detection (the car exterior environment) is different than the signal that is ultimately presented to the user (either the siren, or a statement that an emergency vehicle is coming from a different direction). EX1001, 5:19-38.

58. In another embodiment of the ‘587 patent, the Hearable system, the same clear distinction is made between the signature that is used for detection (i.e., the wearer’s voice in a background noisy environment) and the signal that is ultimately presented to the user (a combination of the wearer’s voice as received by the external microphone and the wearer’s voice as received by the ear canal). EX1001, 5:59-6:17.

59. In still another embodiment of the ‘587 patent, the Earbliss system, again the ‘587 patent distinguishes between the signature that is used for detection (i.e., the snoring of a third party) with the signal that is ultimately presented to the user (the ambient sound environment with the snoring filtered out of the signal). EX1001, 7:62-8:31.

60. Finally, in the Hear and Tell embodiment, the user on detection of a sonic signature, has at their disposal a number of options, including boosting the sound, lowering the sound, attenuating the sound, or replacing the detected sound with an alternative sound to name. EX1001, 9:11-27.

61. As such, it is my opinion that the references in Ground 1 do not teach “sending a portion of a microphone signal to the audio buffer,” and therefore Petitioners have not shown that claim 1 of the ‘587 patent is unpatentable.

2. **Fiedler Does Not Disclose “Sending A Modified Microphone Signal To The Data Buffer...”**

62. Petitioner’s limitation [1.13] of the ‘587 patent recites “sending a modified microphone signal to the data buffer replacing or adding to previous data stored in the data buffer.” It is my understanding that Petitioner and Mr. Schmandt rely entirely on the Fiedler reference for its argument that this limitation is shown in the prior art. Pet. at 34-38.

63. Petitioner and Mr. Schmandt argue that the “modified microphone signal” of the ‘587 patent is disclosed by Fiedler’s deletion of audio data below a certain amplitude threshold prior to capture, and its replacement with a silence node. *Id.* at 36 (citing EX1002, ¶ 192). However, a person of ordinary skill in the art would understand that this does not create a “modified microphone signal.”

64. The system of Fiedler provides a “noise gating” function. That is, after checking to see whether an incoming microphone signal “sample” is above or below a defined threshold, Fiedler’s system either destroys the “samples” (if the amplitude of the sample is below the defined threshold) or does not change them in any way (if the amplitude is above the threshold). EX1007, 5:42-47.

65. In contrast, the embodiments disclosed in the ‘587 patent all perform some operation on the microphone signal. For example, the ‘587 patent discloses that modifications of the signal can include “passing the detected signal through to the user, boosting the sound such that the user is made aware of the detected sound,

rejecting the sound so that the user does not hear the detected sound, attenuating the detected sound, and replacing the detected sound with an alternative sound to name.”

EX1001, 9:21-27.

66. Another embodiment of the ‘587 patent similarly discloses amplifying the detected signal so as to make the wearer of the earpieces aware of the signal’s detection and “eliminate what does not need to be heard while allowing the user to be aware of other sounds in a normal context,” 8:24-31,

67. In yet another embodiment of the ‘587 patent, a system is disclosed that amplifies a signal that is exterior to an automobile so that it can be heard clearly inside the automobile, with additional instructions optionally provided. (*id.*, 5:19-38).

68. In yet another embodiment, the ‘587 patent discloses modifying the voice of a user for clarity and recognizability by both the user and a listener (6:52-7:17).

69. As such, it is my opinion that the Fiedler reference does not teach “sending a modified microphone signal to a data buffer...,” as required by claim 1 of the ‘587 patent.

3. **Claims 3, 4 and 12**

70. Claims 3, 4 and 12 each depend from claim 1. For each of these claims, Petitioner relies on the same teachings of Fielder for its argument that the ‘587

patent's limitations "sending a portion of the microphone signal to the audio buffer" and "sending a modified microphone signal to the data buffer replacing or adding to previous data stored in the buffer" are shown in the prior art. Pet. at 42. For the reasons set forth in Sections VII.A.1.-2. above, Petitioner has failed to show by a preponderance of the evidence that claim 3, 4 and 12 of the '587 patent are unpatentable as well.

B. Ground 1B: Fiedler, Broussard and Victorian Do Not Render Claims 5-9 of the '587 Patent Obvious

71. Ground 1B alleges that the combination of Fiedler, Broussard and Victorian renders claims 5-9 obvious. However, Ground 1B relies on the same arguments as Ground 1A for the term "sending a portion of the microphone signal to the audio buffer." That is, Petitioner does not rely on Victorian to supply these missing limitations. Pet. at 52. Therefore, for the reasons I explained in Section VII.A. above that the combination of Fiedler and Broussard do not disclose the above limitations, Petitioner has failed to show that the addition of Victorian as a reference in Ground 1C renders claim 8 of the '587 patent unpatentable.

C. Ground 1C: Fiedler, Broussard, Victorian and Andreasson Do Not Render Claim 8 of the '587 Patent Obvious

72. Ground 1C alleges that the combination of Fiedler, Broussard, Victorian and Andreasson renders claim 8 obvious. However, Ground 1C relies on

the same arguments as Ground 1A for the term “sending a portion of the microphone signal to the audio buffer” and “sending a modified microphone signal to the data buffer....” That is, Petitioner does not rely on Victorian or Andreasson to supply these missing limitations. Pet. at 52. Therefore, for the reasons I explained in Section VII.A. above that the combination of Fiedler and Broussard do not disclose the above limitations, Petitioner has failed to show that the addition of Victorian and Andreasson as references in Ground 1C renders claim 8 of the ‘587 patent unpatentable.

D. Ground 1D: Fiedler, Broussard and Couper Do Not Render Claims 10 and 11 of the ‘587 Patent Obvious

73. Ground 1D alleges that the combination of Fiedler, Broussard and Couper renders claims 10 and 11 obvious. However, Ground 1D relies on the same arguments as Ground 1A for the terms “sending a portion of the microphone signal to the audio buffer” and “sending a modified microphone signal to the data buffer....” That is, Petitioner does not rely on Victorian or Andreasson to supply these missing limitations. Pet. at 57-60. Therefore, for the reasons I explained in Section VII.A. above that the combination of Fiedler and Broussard do not disclose the above limitations, Petitioner has failed to show that the addition of Couper as a reference in Ground 1D renders claims 10 and 11 of the ‘587 patent unpatentable.

**E. Ground 2: Fiedler, Broussard and Bulhuis Do Not Render
Claims 1 and 2 of the ‘587 Patent Obvious**

74. Ground 2 alleges that the combination of Fiedler, Broussard and Bulhuis render claims 1 and 2 obvious. However, Ground 2 relies on the same arguments as Ground 1A for the terms “sending a portion of the microphone signal to the audio buffer” and “sending a modified microphone signal to the data buffer....” Pet. at 60, 66. That is, Petitioner does not rely on Bulhuis to supply these missing limitations. Therefore, for the reasons I explained in Section VII.A. above that the combination of Fiedler and Broussard do not disclose the above limitations, Petitioner has failed to show that the addition of Bulhuis as a reference in Ground 2 renders claims 1 and 2 of the ‘587 patent unpatentable.

I declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true, and that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both under Section 1001 of Title 18 of the United States Code and that any such willful false statements may jeopardize the validity of the patent at issue in this proceeding.

Dated: December 6, 2024

A handwritten signature in black ink, appearing to read "Eric Tarr", written over a horizontal line.

Dr. Eric Tarr

EXHIBIT A

Eric W. Tarr, Ph.D.

CONTACT INFORMATION	1900 Belmont Blvd. Nashville, TN 37212	Office: 615-460-5722 E-mail: eric.tarr@belmont.edu
ACADEMIC EMPLOYMENT	Associate Professor of Audio Engineering Technology Belmont University, Curb College of Entertainment and Music Business	2019-present
	Assistant Professor of Audio Engineering Technology	2013-2018
EDUCATION	The Ohio State University , Columbus, Ohio USA Ph.D., Electrical and Computer Engineering, 2013 • Advisers: Professor Ashok Krishnamurthy & Professor Susan Nittrouer • Area of Study: Digital Signal Processing/Communication • Minor Area of Study: Control Systems M.S., Electrical and Computer Engineering, 2010 • Advisers: Professor Ashok Krishnamurthy & Professor Susan Nittrouer • Area of Study: Digital Signal Processing/Communication B.S., Electrical and Computer Engineering, 2008 • Specializations: Circuit Design, Control Systems, DSP/Communication Capital University , Columbus, Ohio USA B.A., Dual degree: Mathematics/Engineering, 2008 • Minor: Music (emphasis on Music Technology courses) • Completed advanced music recording program at the Recording Workshop	
EXPERT LITIGATION CONSULTING	Paranjpe, Mahadass, Ruemke LLP , Expert Witness 2021-2023 Provided technical analysis and acoustic testing of Apple Air Pod Pro headphones as well as source code review of iOS and Bluetooth software • <i>Gordoa v. Apple Inc</i>; Case No. 3:22-CV-02900-JSC; United States District Court, Northern District of California – Retention by Plaintiff Quandary Peak , Expert Analysis Consultant 2019-2020 Provided technical analysis of software & hardware evidence • <i>Allscripts Healthcare, LLC v. Andor Health, LLC</i>; Case No. 1:21-cv-00704-MAK; United States District Court, District of Delaware – Retention by Defendant • <i>Truinject Corp v. Nestle Skin Health, S.A.</i>; Case No. 1:19-cv-00592-LPS-JLH; United States District Court, District of Delaware – Retention by Plaintiff • <i>Marina Christensen v. California State University System</i>; Retention by Plaintiff; resolved in arbitration Fragomen, Del Rey, Bernsen & Loewy, LLP , Expert Consultant 2019 Consultant in H1B immigration visa case for software developer position	
AUTHORED TEXTBOOKS	Tarr, E. (2018) <i>Hack Audio: An Introduction to Computer Programming and Digital Signal Processing in MATLAB</i> . • Co-published by the Audio Engineering Society, Focal Press: AES Presents Co-author (2023) <i>Distortion in Music Production: The Soul of Sonics</i> . • Author of Chapter 2: The Development of Audio Software with Distortion	

RESEARCH GRANTS	Unreal Engine Epic Games Grant	2019-present
	- Funding to establish the Laboratory of Audio for Interactive Media at Belmont - Supported graduate student research appointments	
	Tennessee Entertainment Commission Digital Media Grant	2019-2023
	- Funding to develop educational resources for VR and AR systems - Offered free, in-person workshops for people in Middle Tennessee Summer '22, '23	
	AFWERX Air Force Innovation Grant	2019-2020
INDUSTRY EMPLOYMENT	Subject Matter Expert, Aviation Resources and Consulting Services LLC (ARCS)	
	Department of Defense (DoD) grant for using augmented reality technology in air force maintenance and repair training.	
	Gibson Foundation Les Paul Music Innovation Award	2015-2016
	Principal Investigator, "Blockchain Technology in the Music Industry"	
	- Created a virtual currency for peer-to-peer transactions - Developed a method of encoding audio files to be embedded in Smart Contracts	
	Empirical Labs Inc. DSP Contractor	2019-present
	Individually programmed the BIG FrEQ software plug-in	
	Used on hundreds of records and live performances by many top touring artists	
	Impact Soundworks DSP Contractor	2021-present
	Created a software version of the Bogner Ecstasy Guitar Amp	
	KIT Plugins Software Developer	2022-present
	Created the Blackbird A5 plug-in	
	A software re-creation of the API console at Blackbird Studios	
	Density Inc. Software Developer	2020-2024
	Software developer on a team creating a Digital Audio Workstation for DJs	
	ReLab Development DSP Engineer	2021-2023
	Created the machine learning analysis process to model the Maselec MEA-2 EQ	
	Skywalker Sound Ltd. of Lucasfilm Software Consultant	2019-2020
	Created software to automate the process of editing a feature film for television	
	Developed algorithm to change the playback speed of audio without pitch artifacts	
	Used extensively by Disney to prepare movies for television release	
	Apogee Electronics Corp. Software Developer	2018-2019
	Digital modeling of analog circuits, created Opto-3A plug-in	
	Sennheiser Electronics GmbH , Software Development Partner	2018
	Developed augmented-reality iOS app, Impaired Hearing Simulator (IHSIM) , which simulates the listening experience of hearing aids and cochlear implants	
	L.R. Baggs , DSP Contractor	2018-2019
	Adaptive filter design, impulse response modeling	
	Art+Logic , DSP Consultant	2018-2019
	Room response modeling and correction for Gibson/KRK loudspeakers	
	Created machine learning algorithm to automatically adjust speaker performance	
	Soundslates Ltd. , Software Contractor	2018
	Implemented software synthesizer with 20+ virtual analog models of filters	
	Hack Audio, LLC	2015-present
	Founder and Content Creator	
	- Educational resources on DSP, computer programming, and audio - AAX, VST, AU plug-ins	

Computer Programming for Audio Engineers (3 credit hours)	AET 3220
Introduction to the use of computer programming in audio engineering	
• Concepts: variables, data types, block diagrams • Constructs: loops, conditional statements, functions	
Digital signal processing and algorithm development	
• Linear time-invariant systems, convolution, filtering, echo, reverberation • Distortion, amplitude modulation, panning, fades, compressors, expanders	
Audio Software Development (3 credit hours)	AET 4895/5895
C++ programming and JUCE framework	
• Object-oriented programming, real-time signal processing • VST, AU, and AAX plug-in formats, iOS and Android Apps • Circular delay buffers, synthesis from instantaneous phase, ADSR envelopes • GUI parameter control with button, knob, and fader objects, metering	
Digital Audio, Comp. Prog., Signal Analysis (3 credit hours)	AET 5410
Graduate course on programming with MATLAB	
• Mixed-signal systems: sampling, quantization, analog and digital audio • Signal synthesis, measurements of peak, RMS, crest factor, and pitch detection • Graphical User Interface design for audio experiments	
Spectral Analysis: frequency and time-frequency representations of signals	
• Auto- and Cross-correlation, Discrete Fourier Transform, STFT, spectrogram • Complex exponentials, magnitude, phase	
Audio Signal Processing (3 credit hours)	AET 5420
Graduate course on designing audio effects processors	
• Non-linear distortion effects, even and odd functions • Feed-forward, feed-back, bi-quadratic filters • Modulation effects, envelope effects, dynamic range processors	
Spectral analysis of audio effects	
• Impulse response, step response, z-transform • Pole-zero analysis and design of LTI systems	
Transform-based Signal Processing (2 credit hours)	AET 5895
Advanced topics graduate course on signal processing in the spectral domain	
• Overlap-add, overlap-save, uniform and non-uniform partitioned convolution • Frequency-domain filter design, Noise reduction via spectral subtraction • Phase vocoder algorithms, Gabor Transform, Constant-Q Transform and Wavelets	
Signal Processing for Time-Variant Systems (2 credit hours)	AET 5895
Advanced topics graduate course on signal processing	
• Systems with a modulated time delay, Circular buffers and fractional delay • Linear, Sinc, and Lagrange Interpolation, Farrow Structures	
Analog Signal Processing, Digital Modeling (3 credit hours)	AET 5895
Advanced topics graduate course on circuit analysis and simulation	
• Ohm's Law, KVL, KCL, Nodal Analysis, Passive & Active Amplifiers, Op-amps • Laplace Transform, Linear Filters, Bilinear Transform for discretization • Modified Nodal Analysis, State-space Analysis, Discrete Kirchhoff (DK) Analysis • Nonlinear circuits, diodes, transistors, and tubes • Newton-Raphson solver for implicit equations	
Research Design and Methods (3 credit hours)	AET 5500
Graduate course on experimental methods and technical writing	
• Inferential statistics, hypothesis testing, correlation, t-test, analysis of variance • Effect size, Detection Theory, Receiver Operating Characteristic • Technical writing for graduate theses and journal articles • Oral presentations and public speaking on technical topics	

PUBLICATIONS

- Hemami, H., Tarr, E., Li, B., Krishnamurthy, A., Clymer, B., & Dariush, B. (2016). Towards a Cybernetic Model of Human Movement. **Mech. Eng. Res.** 6.
- Lobel, N., Tarr, E., & Bulla, W. (2015). Listener discrimination of high-speed digitization from analog tape masters with spectral matching. **AES** 139.
- Nittrouer, S., Tarr, E., Wucinich, T. & Moberly, A. (2015). Measuring the effects of spectral smearing and enhancement on speech recognition in noise for adults and children **JASA** 137, 2004-2014.
- Tarr, E., Howard, J., & Stager, B. (2014). Listener preferences for analog and digital summing based on music genre. **AES** 137, Paper 9136.
- Moberly, A., Lowenstein, J., Tarr, E., Caldwell, A., Welling, D., Shahin, A., & Nittrouer, S., (2014). Cochlear implant users rely on different acoustic cues for phoneme perception than normal-hearing adults. **J. Speech, Hear. Res.**
- Nittrouer, S., Tarr, E., Boster, V., Cadwell, A. Moberly, A. & Lowenstein, J., (2014). Low-frequency signals support perceptual organization of implant-simulated speech for adults and children. **International J. of Audiology**.
- Nittrouer, S., Lowenstein, J. H., Wucinich, T. & Tarr, E. (2014). Benefits of preserving stationary and time-varying formant structure in alternative representations of speech: Implications for cochlear implants. **JASA** 136, 1845-1856.
- Tarr, E. & Nittrouer, S. (2013). Explaining coherence in coherence masking protection for adults and children. **JASA** 133, 4218-4231.
- Nittrouer, S., Lowenstein, J., & Tarr, E., (2013). Amplitude rise time does not cue the /ba/-/wa/ contrast for adults or children. **J. Speech, Hear. Res.**
- Nittrouer, S., Caldwell, A., Tarr, E., & Lowenstein, J., (2013). Improving speech-in-noise recognition for children with hearing loss: Potential effects of language abilities, binaural summation, and head shadow. **International J. of Audiology**.
- Lowenstein, J., Nittrouer, S., & Tarr, E., (2012). Children weight dynamic spectral structure more than adults: Evidence from equivalent signals. **JASA**, 132, EL443-EL449.
- Nittrouer, S., Caldwell, A., Lowenstein, J., Tarr, E., & Holloman, C., (2012). Emergent literacy skills of kindergarten children with cochlear implants. **Ear & Hearing**.
- Nittrouer, S. & Tarr, E. (2011). Coherence masking protection for speech and non-speech signals in children and adults. **Atten. Percept. Psychophys.**, 73, 2606-2623.
- Tarr, E. & Nittrouer, S. (2011). Coherence masking protection for mid-frequency formants by adults and children **JASA**, 130, EL290-EL296.

CONFERENCE PRESENTATIONS AND POSTER ABSTRACTS

- Tarr, E. (2021). Teaching audio signal processing using a digital audio workstation as the calculator. **AES Conference on Audio Education**.
- Tarr, E. (2019). A program-dependent digital compressor for auditory prostheses specialized for signal transients. **ASA** 178.
- Miller, R., Bulla, W., & Tarr, E. (2018). Detection of the effect of window duration in an audio source separation paradigm. **AES** 147.

Carei, B. & Tarr, E. (2018). Real-time simulation of hearing loss and auditory prostheses. **International Hearing Aid Research Conf.**

Tarr, E. (2017). An open-source audio software plug-in and mobile app for cochlear implant simulation. **Conf. Cochlear Implant Research, U-Wisconsin, Madison.**

Tarr, E. (2016). Introducing programming through acoustics and audio at Belmont University. **ASA 172**, 4pEDa7.

Lowenstein, J.H., Tarr, E., & Nitttrouer, S. (2014). Effects of spectral smearing on sentence recognition by adults and children. **ASA 136**, S2262.

Tarr, E., Krishnamurthy, A., & Nitttrouer, S. (2013). A pitch-synchronous framework for representing, recovering and modifying temporal structure in speech. **CIAP.**

Nitttrouer, S., Tarr, E., Lowenstein, J., & Bolster, V. (2013). Perceptual benefit of a low-frequency acoustic signal with electric stimulation for children and adults. **CIAP.**

Tarr, E., Caldwell, A., Sansom, E., & Nitttrouer, S. (2012). Children with hearing aids show diminished spatial masking release. **IHCON.**

GRADUATE RESEARCH

Tarr, E., *Processing perceptually important temporal and spectral characteristics of speech*. Doctoral dissertation, The Ohio State University, 2013.

Tarr, E., *Formant narrowing using linear predictive coding to improve phonetic perception in children*. Master's thesis, The Ohio State University, 2010.

CONTINUED EDUCATION

Stanford University, CCRMA Summer Workshop Certifications 2014-2019

- *Audio Plug-ins Designed with FAUST*, synthesis and audio effect processors
- *Music Information Retrieval*, machine learning, pattern recognition, Python
- *Designing Musical Games*, Pure Data, Unreal Engine, Unity
- *Virtual Analog Models of Audio Circuitry*, wave digital filters, state-space models
- *Perceptual Audio Coding*, DSP, psychoacoustics, programming of MP3 algorithm
- *Embedded DSP with Faust*, programming microcontrollers, electronics