

**IN THE UNITED STATES PATENT AND TRADEMARK OFFICE**

In re Patent of: Do Hui Teh, et al.

U.S. Patent No.: 6,339,757

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Title: BIT ALLOCATION METHOD FOR DIGITAL AUDIO SIGNALS

**DECLARATION OF DR. CHRIS KYRIAKAKIS**

I hereby declare that all statements made of my own knowledge are true and that all statements made on information and belief are believed to be true. I further declare 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 the Title 18 of the United States Code.

Dated: 01/08/2025

By: 

Chris Kyriakakis, Ph.D.

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I, Chris Kyriakakis of Los Angeles, California, declare that:

**I. ASSIGNMENT**

1. I have been retained on behalf of Dolby International AB (“Dolby”) to offer technical opinions related to U.S. Patent No 6,339,757 (“the ’757 patent”) (Exhibit 2001). I understand that Roku, Inc. (“Roku”) has filed a petition (“the Petition”) requesting that the Patent Trial and Appeal Board (“PTAB” or “Board”) institute an *inter partes* review (“IPR”) proceeding of the ’757 patent.

2. I have been asked to provide my independent analysis of the ’757 patent based on the prior art publications cited in this declaration.

3. I am not and never have been, an employee of Dolby. I received no compensation for this declaration beyond my normal hourly compensation based on my time actually spent analyzing the ’757 patent, the prior art publications cited below, and issues related thereto, and I will not receive any added compensation based on the outcome of any IPR or other proceeding involving the ’757 patent.

**II. QUALIFICATIONS**

4. I am over the age of 18 and am competent to write this declaration. I have personal knowledge, or have developed knowledge of these technologies based upon education, training, or experience, of the matters set forth herein.

5. I have summarized in this section my educational background, career history, publications, and other relevant qualifications.

6. I earned my Bachelor of Science degree in Engineering and Applied Science from the California Institute of Technology (“Caltech”) in 1985. I received my Master of Science degree in Electrical Engineering in 1987 and my Ph.D. in Electrical Engineering in 1993, both from the University of Southern California (“USC”).

7. I am a tenured Associate Professor in the Department of Electrical and Computer Engineering at USC.

8. I am the founding Director of the USC Immersive Audio Laboratory with facilities for experimental work in room acoustics, multichannel audio, and psychoacoustics. This laboratory also serves as a unique teaching facility for my undergraduate and graduate courses in Digital Audio, Acoustics, and Immersive Audio Signal Processing. All courses include modules on the fundamentals of digital audio and how it is converted to digital and back to analog, how it stored, and how it is transmitted with different encoding schemes. Another major course component is Psychoacoustics that focuses on aspects of masking, critical bands, audibility thresholds, and how they are used in audio compression. The graduate course was developed through a two-year grant I received from the National Science Foundation entitled “Collaborative Learning in Engineering Using Immersive Environments,” and was the first of its kind to assess the impact of audio immersion in student learning.

9. In addition to the courses I teach, I have also supervised and served on Ph.D. dissertation committees for more than 30 students.

10. My research interests lie at the intersection of acoustics, psychoacoustics (the science that studies human perception of sound), and audio signal processing. I have worked in the area of audio signal processing for more than 20 years with particular emphasis on spatial and 3D audio. I have published nearly 100 technical papers, including several peer-reviewed papers. I have published a book on spatial audio entitled *Immersive Audio Signal Processing* and hold several patents in audio processing for headphone optimization, acoustic measurement of loudspeakers in rooms, loudspeaker crossover optimization, and loudspeaker response correction using signal processing. My publications examine various aspects of sound measurement, echo cancellation, microphone arrays, noise removal, filter banks, how sound interacts with the acoustical elements of the environment, novel methods for surround sound recording and reproduction, and the perception of sound by human listeners.

11. A major focus of my research has been on developing novel multichannel audio capture and rendering algorithms. In addition to the mathematical development of the algorithms, I have performed several multichannel recordings with multiple wired and wireless microphones over the past 20 years. For example, one of the first ones was in 2003 with a 64-track

recording of a live performance of the group Duran Duran, 24-track recordings of the Boston Symphony and the Handel and Haydn Society, multi-track recordings of experimental music in the WGBH studios in Boston, 48-track recordings of the Dizzy Gillespie band at USC, live multi-track performances of Byzantine chanters in simulated ancient acoustic environments, as well 32-track Byzantine studio chant recordings of Cappella Romana. In all of these recordings, a combination of wired and wireless microphones were used in conjunction with digital multitrack recording gear and SMPTE timecode synchronization. Several of these recordings have been used to compare the performance of digital audio compression methods to better understand how to reduce data rates for streaming over networks.

12. Another focus of my early research was on head-related transfer functions (HRTFs) and how they can be measured and modeled using various signal processing approaches. I developed several implementations of HRTF-based 3D audio for rendering spatial sounds over headphones. I have also researched microphone arrays for applications in enhanced far field voice recognition. This includes methods for Direction of Arrival finding and beam forming from linear and circular microphone arrays. I have also investigated methods for noise removal and reverberation reduction from microphone signals. In addition, over the past few years year I have worked on methods for echo cancellation for applications in wearable devices. Previous other work includes

loudspeakers and how they interact with their acoustical environment. In particular, I have researched the role of sound reflections, absorption, and diffusion in the performance of loudspeakers. I have also published several technical papers on acoustical measurement methods in rooms and device enclosures and developed novel signal processing algorithms for optimizing sound system performance. Other topics I have researched include multichannel audio acquisition and rendering, virtual microphones and virtual speakers, hybrid headphone-loudspeaker rendering methods, and advanced signal processing techniques for optimizing sound quality from small portable devices.

13. From 2003-2018, I was the co-founder and Chief Technology Officer of Audyssey Laboratories, a USC spin-off company that develops and licenses audio technology to leading consumer electronics and automotive companies around the world. As part of my work at Audyssey, I developed a measurement methodology for headphones and earphones and created a database with 500+ measurements of products from nearly 100 manufacturers, created audio algorithms related to sound optimization that are used in millions of home theater systems, cars, and IMAX theaters world-wide.

14. I also developed audio algorithms and designed speakers. These speakers were novel acoustical designs that used a combination of unique enclosures, passive radiators (i.e., speaker diaphragms that are driven by changes

in surrounding air pressure), and audio signal processing to optimize their performance and overcome limitations that arise from small drivers and enclosures. For example, we used signal processing technologies combined with novel acoustical design to extend the bass response of small woofers and passive radiators beyond what was previously possible in small speaker enclosures. The innovations in these designs have received awards, including *Popular Science's* "Best of What's New."

15. In 2019, I joined Syng as the Chief Audio Scientist. Syng creates innovative products for next generation spatial audio and my role there includes research and development of audio algorithms for sound capture using microphone arrays to measure the acoustics of the room and loudspeaker beamforming for wirelessly rendering sound in various directions.

16. In April 2018 I was elected as Senior Member of the Institute of Electrical and Electronic Engineers ("IEEE") in recognition of my contributions to the field of engineering. I am also a member of the Audio Engineering Society (AES), an association for professionals in the audio industry.

17. In 2006, I received a World Technology Network Award. This organization presents awards to innovators in several areas in which technology can foster a paradigm change. My award was for innovations in immersive audio that enable new capabilities in media and journalism. Other award recipients at

that event included Vice President Al Gore, Google, and Space-X. In addition, my work has been featured in multiple news articles, including pieces by *The Atlantic*, the *BBC World Service*, the *New York Times*, and National Public Radio.

18. In the late 1990s and early 2000s, I was a faculty researcher and later Deputy Director of the National Science Foundation's engineering research center established at USC. I studied the fundamental and technological limitations of immersive audio and the role of acoustics on the performance of microphones and loudspeakers. In 2003, together with one of my graduate students, I received the award for Best Paper at the IEEE Conference on Signals, Systems and Computers.

19. My curriculum vitae, attached as Exhibit A, includes a list of publications on which I am a named author. It contains further details regarding my experience, education, publications, and other qualifications provide to an expert opinion in connection with this proceeding.

### **III. SUMMARY OF CONCLUSIONS FORMED**

20. This Declaration explains the conclusions that I have formed based on my analysis. To summarize those conclusions:

- **Ground 1:** Based upon my knowledge and experience and my review of the prior art publications in this declaration, I believe that the combination of Mahieux and Fielder as set forth in Ground

1 of the Petition fails to render obvious claims 1-3, 6-7, 10-11, 14, and 16 of the '757 patent.

- **Ground 2:** Based upon my knowledge and experience and my review of the prior art publications in this declaration, I believe that the combination of Mahieux, Fielder, and Akagiri as set forth in Ground 2 of the Petition fails to render obvious claims 1-17 of the '757 patent.
- **Ground 3:** Based upon my knowledge and experience and my review of the prior art publications in this declaration, I believe that the combination of Veldhuis and Galand as set forth in Ground 3 of the Petition fails to render obvious claims 1-3, 6-7, 10-11, 14, and 16 of the '757 patent.
- **Ground 4:** Based upon my knowledge and experience and my review of the prior art publications in this declaration, I believe that the combination of Veldhuis, Galand, and Reaves as set forth in Ground 4 of the Petition fails to render obvious claims 1-17 of the '757 patent.

#### **IV. LEVEL OF ORDINARY SKILL IN THE ART**

21. I have been informed that a person of ordinary skill in the art (“POSITA”) is a hypothetical person who is presumed to have the skill and

experience of an ordinary worker in the field at the time of the alleged invention.

The '757 patent was filed December 6, 1993, and claims priority to a Japanese patent application filed February 19, 1993 ("Critical Date"). Because I do not know at what date the invention as claimed was made, if ever, I have used the Critical Date of the '757 patent as the point in time for claim interpretation purposes. My opinion does not change if the invention date is earlier.

22. The Petition proposes that a POSITA in the field of the '757 patent "would have had a Bachelor of Science (BS) degree in Computer Science, Computer Engineering, Electrical Engineering, or an equivalent field, as well as at least 2-3 years of academic or industry experience in the fields of audio encoding, analog-to-digital signal conversion, or digital signal processing." Pet., Section III (citing EX1003, ¶¶19-24). For purposes of Patent's Owner's preliminary response, I do not challenge that definition.

23. Based on my experiences, I have a good understanding of the capabilities of a POSITA, as proposed in the Petition. Indeed, I have taught, mentored, advised, and collaborated closely with many such individuals over the course of my career. Based on my knowledge, skill, and experience, I have an understanding of the capabilities of one of ordinary skill. For example, from my industry experience, I am familiar with what an engineer would have known and found predictable in the art. From teaching and supervising my post-graduate

students, I also have an understanding of the knowledge that a person with this academic experience possesses. Furthermore, I possess those capabilities myself.

## **V. LEGAL PRINCIPLES**

24. I am not a lawyer and I will not provide any legal opinions in this IPR. Although not a lawyer, I have been advised that certain legal standards are to be applied by technical experts in forming opinions regarding the meaning and validity of patent claims.

### **A. Claim Interpretation**

25. I understand that claim terms are generally given their plain and ordinary meaning based on the patent's specification and file history as understood by a person of ordinary skill in the art at the time of the purported invention. In that regard, I understand that the best indicator of claim meaning is its usage in the context of the patent specification as understood by a POSITA. I further understand that the words of the claims should be given their plain meaning unless that meaning is inconsistent with the patent specification or the patent's history of examination before the Patent Office. I also understand that the words of the claims should be interpreted as they would have been interpreted by a POSITA at the time of the invention was made (not today).

**B. Priority**

26. I understand that a continuation application is a later-filed application that has the same disclosure (specification and figures) as an earlier filed application to which the later-filed application claims priority. A continuation is generally entitled to the same priority date as the later-filed application to which it claims priority.

**C. Anticipation**

27. I understand that a patent claim is invalid as anticipated if each and every element as set forth in the claim is found, either expressly or inherently described, in a single prior art reference. I also understand that, to anticipate, the reference must teach all of the limitations arranged or combined in the same way as described in the claim. I do not rely on anticipation in this declaration.

28. With respect to inherency, I understand that the fact that a certain result or characteristic may occur or be present in the prior art is not sufficient to establish the inherency of that result or characteristic. Instead, the inherent characteristic must necessarily flow from the teaching of the prior art.

**D. Obviousness**

29. I understand that a patent claim is invalid if the claimed invention would have been obvious to a person of ordinary skill in the field at the time of the purported invention, which is often considered the time the application was filed.

Even if all of the claim limitations are not found in a single prior art reference that anticipates the claim, the claim can still be invalid.

30. To obtain a patent, a claimed invention must have, as of the priority date, been nonobvious in view of the prior art in the field. I understand that an invention is obvious when the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art.

31. I understand that, to prove that prior art or a combination of prior art makes a patent obvious it is necessary to: (1) identify the particular references that, singly or in combination, make the patent obvious; (2) specifically identify which elements of the patent claim appear in each of the asserted references; and (3) explain a motivation, teaching, need, market pressure or other legitimate reason that would have inspired a person of ordinary skill in the art to combine prior art references to solve a problem.

32. I also understand that certain objective indicia can be important evidence regarding whether a patent is obvious or nonobvious. Such indicia include:

- Commercial success of products covered by the patent claims;
- A long-felt need for the invention;

- Failed attempts by others to make the invention;
- Copying of the invention by others in the field;
- Unexpected results achieved by the invention as compared to the closest prior art;
- Praise of the invention by the infringer or others in the field;
- The taking of licenses under the patent by others;
- Expressions of surprise by experts and those skilled in the art at the making of the invention; and
- The patentee proceeded contrary to the accepted wisdom of the prior art.

33. To the extent these factors have been brought to my attention, if at all, I have taken them into consideration in providing my opinions and conclusions.

## **VI. MATERIALS CONSIDERED**

34. My analysis and conclusions set forth in this declaration are based on my educational background and experiences in the field (*see* Section II). Based on my knowledge and experience, I believe that I am considered to be an expert in the field. Also, based on my knowledge and experience, I understand and know of the capabilities of persons of ordinary skill in the field during the early 1990s, and I taught, participated in organizations, and worked closely with many such persons in the field during that time frame.

35. As part of my independent analysis for this declaration, I have considered the following: the background knowledge/technologies that were commonly known to persons of ordinary skill in this art during the time before the earliest claimed priority date for the '757 patent; my own knowledge and experiences gained from my work experience in the field of the '757 patent and related disciplines; and my experience in working with others involved in this field and related disciplines.

36. In addition, I have analyzed the following publications and materials:

- U.S. Patent No. 6,339,757 to Teh, et al. (“the ‘757 patent”) (Exhibit 2001) and its accompanying prosecution history (Exhibit 2012),
- U.S. Patent No. 5,185,800 to Mahieux (“Mahieux”) (Exhibit 2003),
- U.S. Patent No. 5,109,417 to Fielder et al. (“Fielder”) (Exhibit 2004),
- U.S. Patent No. 5,241,603 to Akagiri et al. (“Akagiri”) (Exhibit 2005),
- U.S. Patent No. 4,896,362 to Veldhuis et al. (“Veldhuis”) (Exhibit 2006),

- U.S. Patent No. 4,677,671 to Galand et al. (“Galand”) (Exhibit 2007),
- U.S. Patent No. 5,579,431 to Reaves (“Reaves”) (Exhibit 2008),
- U.S. Patent No. 5,105,463 to Veldhuis et al. (“Veldhuis’463”) (Exhibit 2009),
- European Patent Publication No. 0145788A1 to Akagiri (“Akagiri’788”) (Exhibit 2010), and
- European Patent Publication No. 0525809A2 to Akagiri (“Akagiri’809”) (Exhibit 2011).

37. My analysis and conclusions set forth in this declaration are based on the perspective of a POSITA.

## **VII. OVERVIEW OF THE ’757 PATENT**

### **A. Specification**

38. The ’757 patent “relates to the efficient information coding of digital audio signals for transmission or digital storage media especially a quantization method using an efficient bit allocation method for information coding.” Exhibit 2001, 1:7-10.

39. One such method includes “a dynamic bit allocation procedure that is adapted to the human auditory system and yet with a low level of complexity so that high audio quality can be achieved while at the same time meeting the low

cost target.” Exhibit 2001, 2:41-45. An example implementation of this procedure is shown in FIG. 1, reproduced below.

**F i g . 1**

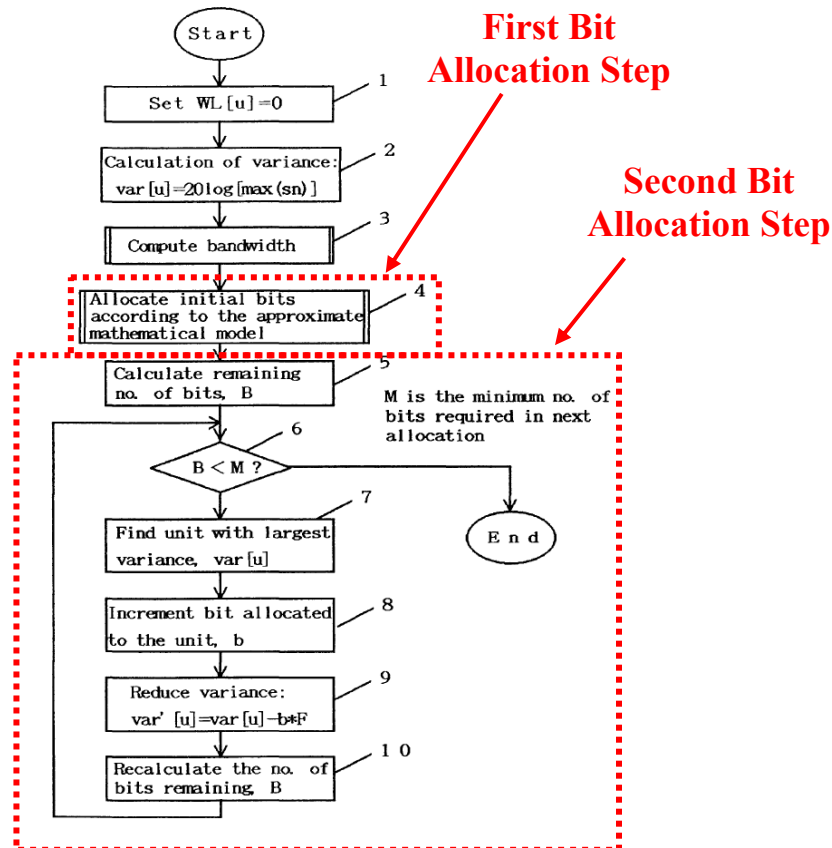


Exhibit 2001, FIG. 1 (annotated)<sup>1</sup>

40. In general, this procedure dynamically allocates bits for quantizing and coding information regarding an audio signal according to at least two different and distinct bit allocation steps. In a first bit allocation step, bits are initially allocated to “frequency-units” that can be heard by humans based on an

<sup>1</sup> Annotations to the figures throughout this Declaration are shown in color.

approximate mathematical model. *See, e.g.*, FIG. 1, operation 4. Frequency-units” that can be heard by humans are also referred to as “valid” or “available” frequency-units. In a second bit allocation step, additional bits are subsequently allocated to frequency-units, focusing on available frequency-units based on psychoacoustic criteria. *See, e.g.*, FIG. 1, operations 5-10.

41. Specifically, according to the dynamic bit allocation procedure, a system obtains an digital audio signal “having a spectral and temporal structure.” Exhibit 2001, Abstract. Further, the system transforms the audio signal using a modified discrete cosine transform (MDCT) into several spectral lines that span the frequency range of the signal. Exhibit 2001, 4:21-24. Each spectral line represents the contents of the audio for each specific frequency. For example, the frequency of each spectral line corresponds to the frequency of a particular modified cosine basis function, and the amplitude of each spectral line corresponds to the amplitude of the audio signal at that frequency (e.g., a coefficient of the modified cosine basis function).

42. Further, the system divides the frequency interval spanning the spectral lines into several groups, an example of which shown in FIG. 6. Exhibit 2001, 4:21-24. These groups are also referred to as “frequency-units” or “units.” *Id.* In this example, the transformed audio signal has a frequency interval that spans 256 spectral lines. This frequency interval is divided into 26 different

“frequency-units.” For example, a first frequency-unit (labeled as “unit 0”) spans eight spectral units (labeled as “1” to “8”), a second frequency unit (labeled as “unit 1”) spans four frequency-units (labeled as “9” to “12”), and so forth.

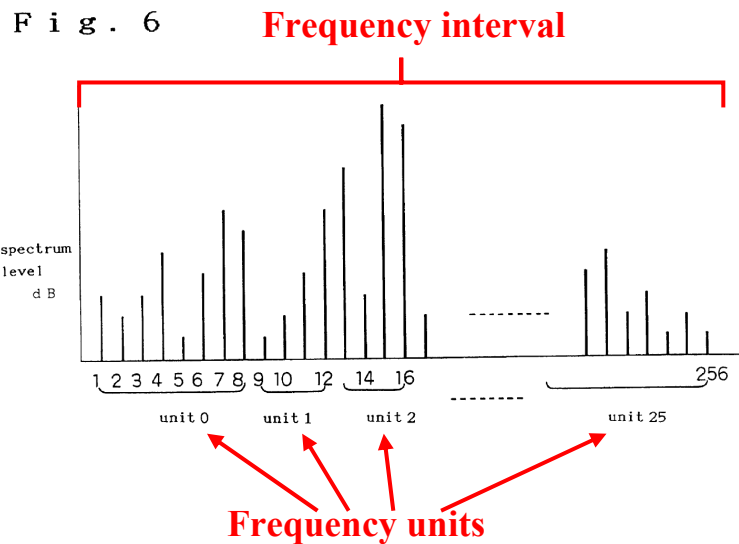


Exhibit 2001, FIG. 6 (annotated)

43. The system dynamically allocates bits for representing the spectral lines of each of the frequency-units according to a multi-step process. An example of this process is shown in FIG. 1.

**F i g . 1**

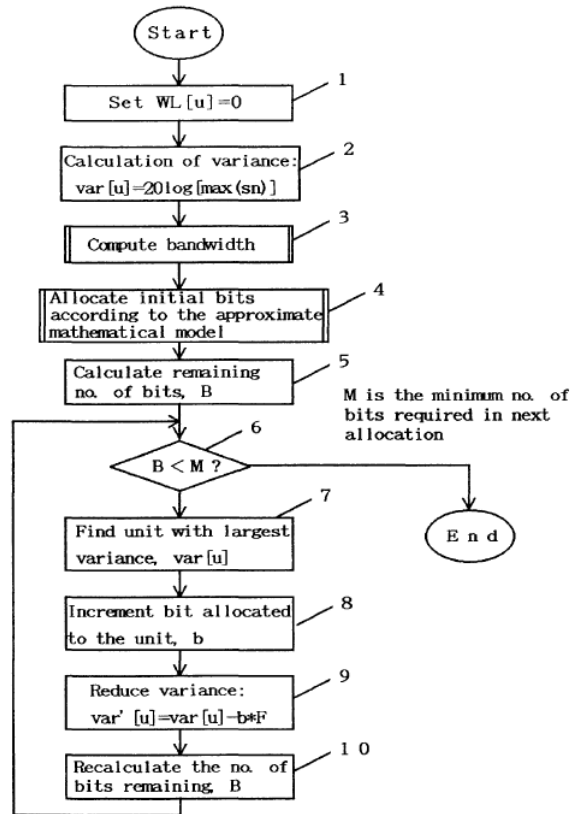


Exhibit 2001, FIG. 1.

44. As a first step, the system sets the bit allocation for each of the frequency-units to zero. Exhibit 2001, 3:47-49, FIG 1 (operation 1). As an example, the system sets a variable “WL[u]” to zero, where “WL” represents the number of bits allocated to a frequency-unit “u.” *Id.* 3:47-49, 7:7-8, FIG. 1.

45. Next, the system determines a “representative” for each of the frequency-units. Exhibit 2001, 3:50-59, FIG 1 (operation 2). In some implementations, a representative of a frequency-unit includes a “variance” of the frequency-unit. Exhibit 2001, 3:50-59. The variance can be calculated by

determining at least one of (i) “the maximum spectral sample amplitude in the unit” or (ii) “an average of the spectral components within the unit.” *Id.* (emphasis added).

46. Next, the system determines a bandwidth of transmission for the audio signal. Exhibit 2001, 3:59-60, 4:29-48, FIG. 1 (operation 3). An example process for determining the bandwidth is described with reference to FIG. 2. Specifically, the system initially sets a bandwidth counter to zero. Exhibit 2001, 4:34-35, FIG. 2, operation 31). The system then increases the bandwidth counter when each of the frequency-units meets certain criteria with respect to a “threshold” corresponding to the limits of human hearing. Exhibit 2001, 4:35-48, FIG. 2 (operations 32-36).

**F i g . 2**

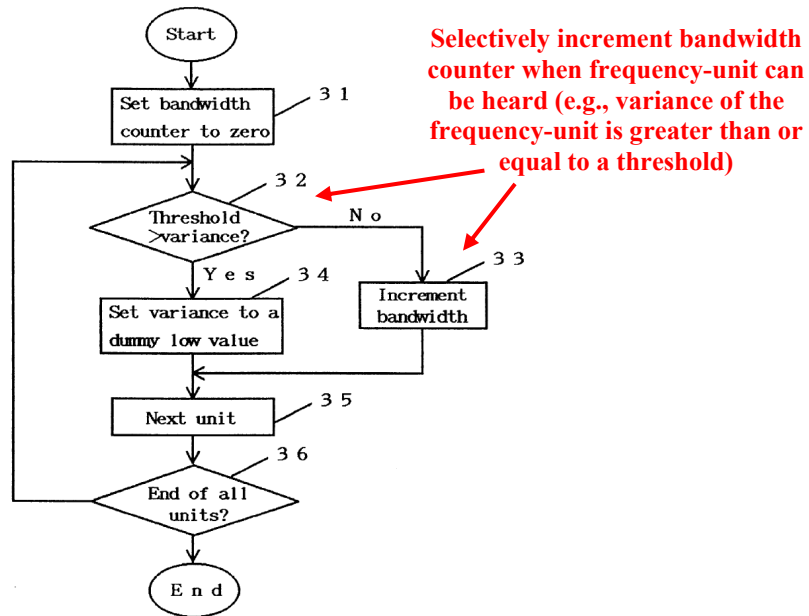


Exhibit 2001, FIG. 2 (annotated)

47. Specifically, the system iterates through each of the frequency-units, and selectively increases the bandwidth counter depending on whether the representative (e.g., variance) of the frequency-unit is greater than or equal to the threshold. For instance, upon determining that the representative of the frequency-unit is greater than or equal to the threshold (indicating that the contents of the audio signal in that frequency-unit can be heard by a human), the system increases the bandwidth counter. Exhibit 2001, 4:35-42, FIG. 2 (operations 32 and 33). In contrast, upon determining that the representative of the frequency-unit is less than the threshold (indicating that the contents of the audio signal in that frequency-unit cannot be heard by a human), the system maintains the bandwidth counter. Exhibit

2001, 4:43-45, FIG. 2 (operations 34 and 35). In this manner, the system counts how many frequency-units are available for bit allocation (e.g., frequency-units that can be heard by a human) to facilitate bit allocation amongst those frequency-units.

48. In particular, the system performs a first bit allocation step in which bits are allocated to frequency-units based on an approximate mathematical model. Exhibit 2001, 3:60-62, 4:49-5:18, FIG. 1 (operation 4). An example mathematical model is represented by equation (1):

$$WL[u] = \frac{\text{var}[u]}{F} - \frac{1}{NF} \sum_{u=0}^N \text{var}[u] + \frac{Ba}{AN} \quad \text{equation (1)}$$

WL[u]: number of bits allocated to unit u  
var[u]: variance of unit u  
F: psychoacoustic weighting factor (a constant)  
N: Number of valid units  
Ba: Number of bits set aside for initial allocation  
A: Approximate average number of spectral samples in a unit

Exhibit 1, Equation (1) (annotated)

49. According to this relationship, “N” represents the counted “[n]umber of valid units” (corresponding to the number of frequency-units that can be heard by a human, and thus are available for bit allocation). Exhibit 2001, 4:62-5:8. This value is expressly used to determine how many bits “WL[u]” to allocate to each of the frequency-units “u,” by virtue of its inclusion in equation (1).

50. Intuitively, equation (1) represents:

- (i) determining a default initial number of bits to allocate to each available (i.e., valid) frequency-unit (represented by the third term, blue), and
- (ii) modifying the initial number of bits that are actually allocated to each of the available frequency-units based on whether the variance of the frequency-unit (red) exceeds the average variance across all of frequency-units (green), normalized by a “psychoacoustic weighting factor,” F.

51. Notably, the system does not initially allocate any bits to frequency-units that cannot be heard by a human (e.g., frequency-units that are below the threshold of hearing), as determined by the bandwidth counting operations described with respect to FIG. 2.

52. The system subsequently performs a second bit allocation step in which additional bits are allocated to the available frequency-units based on psychoacoustic criteria. Exhibit 2001, 3:62-67, FIG. 1 (operations 5-10).

53. Specifically, the system iteratively allocates the bits left over after the first bit allocation (e.g., calculated by operation 5, which is subsequent to operation 4 of FIG. 1) by identifying the frequency-unit having the highest variance, and (i) increasing the bit allocation to that frequency-unit by “b,” and (ii) reducing the

variance of that frequency-unit by  $b \cdot F$ . Exhibit 2001, 4:1-13, FIG. 1 (operations 7-9). Reducing the variance of the frequency-unit lowers the likelihood that the frequency-unit will be allocated additional bits in the next iteration (but does not entirely preclude the allocation of additional bits to the frequency-unit). The amount that the variance is reduced depends on “F” (described as a “psychoacoustic weighting factor”). Exhibit 2001, 4:12-14.

54. This second bit allocation step is iteratively repeated until the number of remaining bits B is less than the minimum number of bits required in the next bit allocation (e.g., indicating that there are an insufficient number of bits remaining to perform any further bit allocations). Exhibit 2001, 4:1-17, FIG. 1 (operation 6).

## **B. Claims**

55. The table below includes identifiers for the '757 patent claims and limitations, which are referenced throughout.

| Claim | Identifier | Claim Language                                                                                                                                                                                                                                                                                              |
|-------|------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| 1     | [1P]       | A method of determining a bit allocation for a quantization of digital audio signals having spectral and temporal structure wherein the digital audio signals are obtained by buffering audio signals in frames and decomposing the digital signals into spectral components, which comprises the steps of: |
|       | [1A]       | a step for dividing at least one frequency interval into a plurality of frequency-units;                                                                                                                                                                                                                    |
|       | [1B]       | a step for obtaining a representative of each frequency-                                                                                                                                                                                                                                                    |

| Claim | Identifier | Claim Language                                                                                                                                                                                                                     |
|-------|------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
|       |            | unit obtained;                                                                                                                                                                                                                     |
|       | [1C]       | a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria;                                                                                                           |
|       | [1D]       | a first bit allocation step for determining an initial bit allocation for each of said available frequency-units by using an approximate mathematical model using said counted number of available frequency-units; and            |
|       | [1E]       | second bit allocation step for determining further bit allocation for each of said available frequency-units based on psychoacoustic criteria.                                                                                     |
| 2     | [2]        | Method according to claim 1, wherein said mathematical model is derived from an in-band masking effect such that an error spectrum will follow more closely to the original signal spectrum.                                       |
| 3     | [3]        | Method according to claim 2, wherein each frequency-unit is represented by a single spectral component.                                                                                                                            |
| 4     | [4]        | Method according to claim 2, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.                                                               |
| 5     | [5]        | Method according to claim 2, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.                                                                         |
| 6     | [6]        | A method according to claim 1, wherein said second bit allocation step is to determine the bit allocation through a psychoacoustically weighted inverse relationship with the said representative of each of said frequency-units. |
| 7     | [7]        | Method according to claim 6, wherein each frequency-unit is represented by a single spectral component.                                                                                                                            |
| 8     | [8]        | Method according to claim 6, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.                                                               |

| Claim | Identifier | Claim Language                                                                                                                                                                                |
|-------|------------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| 9     | [9]        | Method according to claim 6, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.                                    |
| 10    | [10]       | A method according to claim 1, wherein said counted number of available frequency-units is a number of the frequency-units which have the representative exceeding a human hearing threshold. |
| 11    | [11]       | Method according to claim 10, wherein each frequency-unit is represented by a single spectral component.                                                                                      |
| 12    | [12]       | Method according to claim 10, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.                         |
| 13    | [13]       | Method according to claim 10, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.                                   |
| 14    | [14]       | Method according to claim 1, wherein each frequency-unit is represented by a single spectral component.                                                                                       |
| 15    | [15]       | Method according to claim 1, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.                          |
| 16    | [16]       | Method according to claim 1, wherein each frequency-unit is obtained by dividing said spectral and temporal structure with at least one frequency interval.                                   |
| 17    | [17]       | Method according to claim 1, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.                                    |

### C. Prosecution History

56. The '757 Patent was allowed after an appeal from the examiner's prior art rejections to the Board of Patent Appeals and Interferences (BPAI).

Exhibit 2012, 264-266. The examiner indicated that as per the BPAI's Decision on Appeal, the prior art cited during prosecution—in particular, U.S. Patent No. 5,105,463 to Veldhuis et al. ("Veldhuis'463") and U.S. 5,151,941 to Nishiguchi et al. ("Nishiguchi") failed to render obvious "the specific limitations [in the independent claims] pertaining to":

- (i) "a step for counting a number of available frequency units by using the representative, based on the psychoacoustic criteria," and
- (ii) "a first bit allocation step for determining an initial bit allocation for each of said available frequency units by using an approximate mathematical model using said counted number of available frequency-units." Exhibit 2012, 265.

## **VIII. INTERPRETATION OF THE '757 PATENT CLAIMS**

57. I understand that, for purposes of my analysis in this inter partes review proceeding, the terms appearing in a patent claim should be interpreted according to their ordinary and customary meaning of such claim as understood by one of ordinary skill in the art and the prosecution history pertaining to the patent. In that regard, I understand that the best indicator of claim meaning is its usage in the context of the patent specification as understood by a POSITA. I further understand that the words of the claims should be given their plain meaning unless that meaning is inconsistent with the patent specification or the patent's history of

examination before the Patent Office. I also understand that the words of the claims should be interpreted as they would have been interpreted by a POSITA at the time of the invention was made (not today). Because I do not know at what date the invention as claimed was made, if ever, I have used the Critical Date of the '757 patent as the point in time for claim interpretation purposes. My opinion does not change if the invention date is earlier.

## **IX. GROUNDS 1 AND 2**

58. The Petition alleges that (i) claims 1-3, 6-7, 10-11, 14, and 16 are obvious in view of Mahieux and Fielder (Ground 1), and (ii) claims 1-17 are obvious in view of Mahieux, Fielder, and Akagiri (Ground 2). Pet, Sections VI-VII.

59. However, as I explain below, these invalidity grounds fail to establish that the challenged claims are obvious for at least the following reasons.

### **A. Overview of Mahieux**

60. In general, Mahieux describes a device that “enables the coding of digital audio data,” an example of which is shown in FIG. 1. Exhibit 2003, 5:14-16, FIG. 1.

61. During operation, the device receives an “input signal  $x(n)$ ” in the time-domain, and transforms the input signal into the frequency-domain using a “modified discrete cosine transform (MDCT)” process. Exhibit 2003, 5:20-37.

This “transform corresponds to a projection on a cosine base,” and produces “transformation coefficients” representing the contents of the audio signal at each of several different frequencies (e.g., corresponding to different respective cosine basis functions). Exhibit 2003, 5:26.

62. During the coding process, the device eliminates at least some of the coefficients (e.g., if the coefficients are less than a “masking threshold,” indicating that the audio at that frequency is “masked” by more prominent frequencies within the signal, and thus are inaudible). Exhibit 2003, 5:66-6:17, 6:61-8:13, FIG. 2.

63. Further, coefficients are quantized according to different numbers of bits, as calculated by a bit allocation module. Exhibit 2003, 6:28-58, 11:63-12:56, FIGS. 1 and 7.

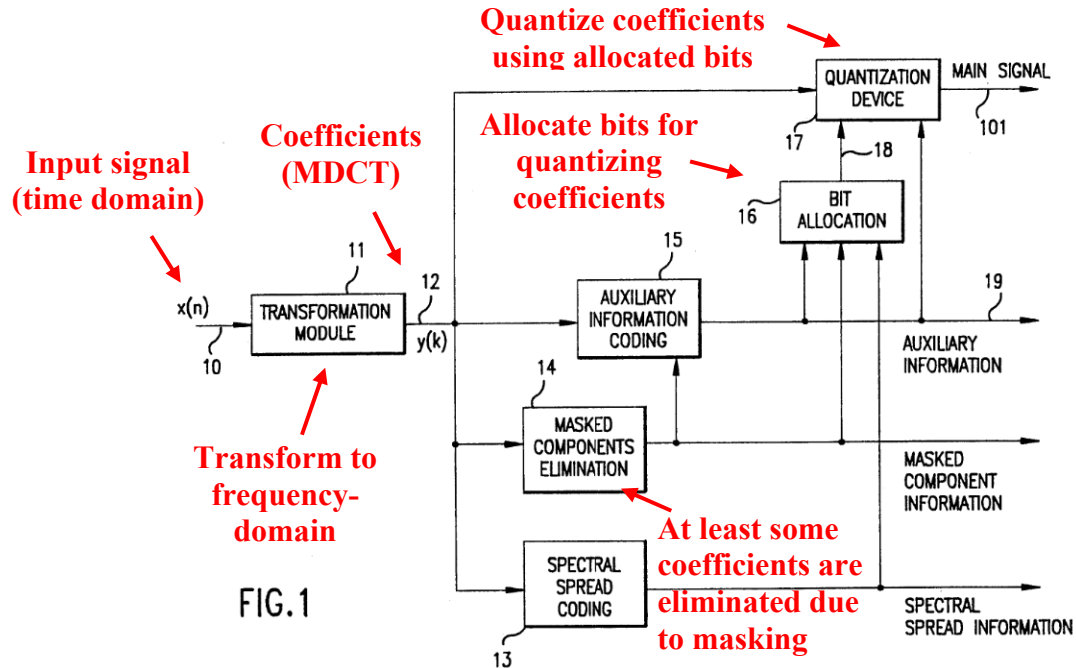


EXHIBIT 2003, FIG. 1 (annotated)

64. The bit allocation module 16 (shown in greater detail in FIG.7) allocates bits “R(k)” to each coefficient “y(k)” based on a specific formula (indicated in block 72 of FIG. 7).

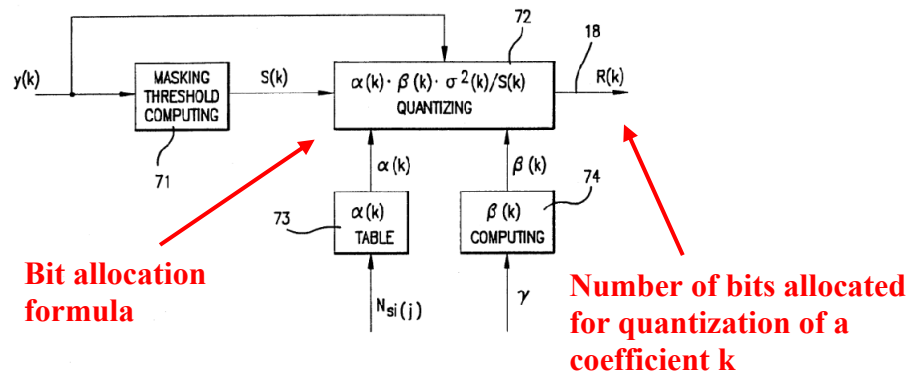


FIG. 7

EXHIBIT 2003, FIG. 2 (annotated)

65. According to this equation:

- $\alpha(k)$  represents “the performance factor of the quantizer to be used for the coefficient  $y(k)$ ,”
- $\beta(k)$  is a function calculated based on “the spectral spread  $\gamma$ ” in order to “force the allocation of the bits to grant a greater number of bits to the zones of the spectrum that contain peaks,”
- $\sigma(k)$  is a “spectrum descriptor,” extended to all of the coefficients  $y(k)$  of [a] band,” and
- $S(k)$  is a “masking threshold.” Exhibit 2003, 12:3-56.

**B. Overview of Fielder**

66. Fielder describes techniques for “high-quality low bit-rate digital signal processing of audio signals, such as music signals.” Exhibit 2004, 1:14-16. Among other techniques, Fielder describes an “adaptive bit allocation” process in

which “for each transform block,” a system “assigns a fixed number of additional bits to specific coefficients in four phases.” Exhibit 2004, 22:57-27:33.

67. The initial phase, “phase zero,” “is an initialization process for the remaining phases.” Exhibit 2004, 23:13-14.

68. In “phase one,” the system “assigns bits, up to a maximum of four per transform coefficient, to the coefficients within the same critical band of those frequency components with the greatest spectral energy. If all allocable bits are assigned during phase one, the allocation process stops.” Exhibit 2004, 23:14-18.

69. “If not, phase two allocates additional bits to the transform coefficients which were allocated bits during phase one such that the total adaptively allocated bits for each coefficient is four. If all allocable bits are assigned during phase two, the allocation process stops.” Exhibit 2004, 23:19-23.

70. “If any bits remain, phase three allocates bits to those coefficients which are adjacent to coefficients that were allocated bits during phase one and two.” Exhibit 2004: 23:23-26.

71. Fielder’s multi-step process for allocating bits, therefore, proceeds as follows. Initially, Fielder’s system prioritizes adaptively allocating bits to coefficients that are in the same “critical band” (e.g., a particular frequency range) as frequency components that have the greatest spectral energy (“phase one”).

Allocation of bits is limited to a maximum of four bits per coefficient. Exhibit 2004, 25:45-68.

72. Next, if any bits remain, Fielder's system adaptively allocates additional bits to the coefficients that were previously allocated bits in "phase one," up to a maximum of four adaptively allocated additional bits per coefficient ("phase two"). Exhibit 2004, 25:26-1:50.

73. Next, if any bits remain, Fielder's system adaptively allocates bits to coefficients that are adjacent to those that were allocated bits in "phase one" and "phase two" ("phase three"). Exhibit 2004, 26:51-25.

**C. [Ground 1] Mahieux-Fielder Does Not Disclose "A Step For Counting A Number Of Available Frequency-Units By Using The Representative, Based On Psychoacoustic Criteria," As Required By The Challenged Claims**

74. Claim 1 recites "a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria." The Petition alleges that Mahieux discloses and/or renders obvious this feature, relying on various operations described in Mahieux. Pet., Section VI.D.4. However, none of these operations amounts to *counting a number of available frequency-units*, as recited in claim 1.

75. In Section VI.D.4, the Petition alleges that Mahieux's "*frequency bands*" correspond to the claimed "frequency-units." Pet., 21 (first full paragraph).

Specifically, the Petition contends that “[t]he resulting output [in Mahieux] is a frequency ‘spectrum’ (claimed ‘frequency interval’) that is ‘cut up’ or ‘divided’ ‘into *frequency bands*’ (claimed ‘*frequency-units*’).” *Id.* (original emphasis modified). See also Exhibit 2003, FIG. 5 (reproduced below with the Petition’s annotations).

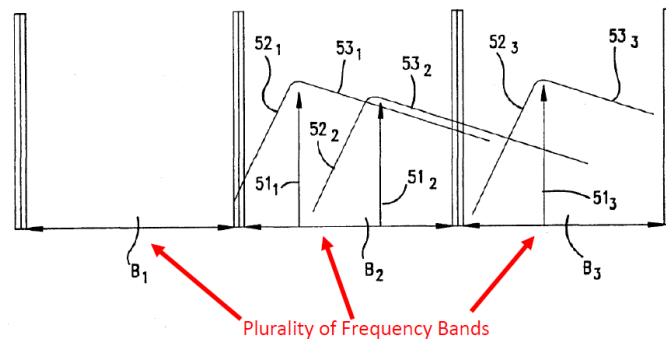


Exhibit 2003, FIG. 5 (as annotated in Pet., 22)

76. In addition, the Petition alleges that Mahieux’s frequency bands are “*available*” if they are “*audible* and, thus, ... are available for bit allocation” (e.g., if the frequency bands have a “power” that is greater than a “masking threshold corresponding to the human auditory system”). Pet., 24 (first full paragraph to second full paragraph) (emphasis added).

77. However, under the Petition’s construction, Mahieux fails to disclose or render obvious *counting* the number of Mahieux’s “frequency bands” that are audible (*i.e.*, the purported “available frequency-units”).

78. Specifically, in Mahieux's system, each "frequency band" includes *several spectral lines*, which are understood to represent the contribution of each spectral component within the frequency band to the overall audio signal. Exhibit 2003, 7:6-15. The relationship between Mahieux's frequency bands and spectral lines are shown in FIG. 5, which depicts example frequency bands B<sub>1</sub>, B<sub>2</sub>, and B<sub>3</sub>, where frequency bands B<sub>2</sub> and B<sub>3</sub> include respective representative spectral lines 51<sub>1</sub>, 51<sub>2</sub>, and 51<sub>3</sub>. The amplitudes of the spectral lines are represented by respective coefficients.

79. Further, Mahieux's system compares *each of the spectral components' coefficients* to a threshold value (represented by the lines 52<sub>1</sub>-52<sub>3</sub> and 53<sub>1</sub>-53<sub>3</sub> in FIG. 5) in order to determine which *individual spectral components* are audible. Exhibit 2003, 7:3-8:9. For example, as Mahieux explains, "[a]s shown in FIG. 5, it would appear that, for each line 51 transmitted, the lines that are beneath a line 52 of -25 dB per critical band upstream of the line and beneath a line 53 of -10 dB per downstream critical band, are inaudible." Exhibit 2003, 7:3-8:9.

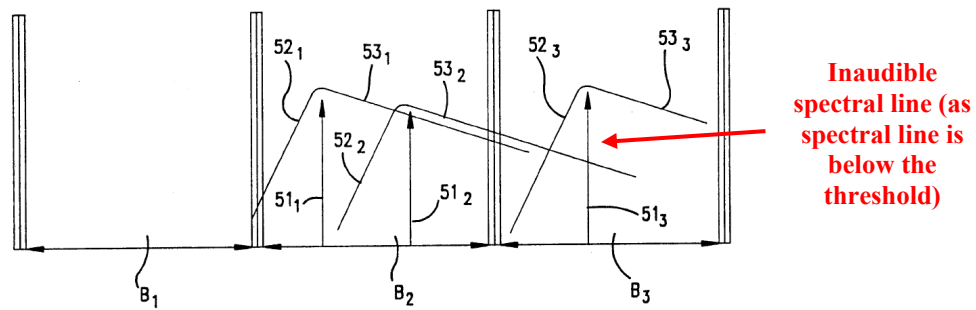


Exhibit 2003, FIG. 5 (annotated)

80. That is, in comparing each *spectral line* to a threshold value, Mahieux's system does not determine whether a "*frequency band*" as a whole is audible (i.e., allegedly "available"). Instead, in making this comparison, Mahieux's system determines whether each of *several constituent spectral components* within the frequency bands are individually audible.

81. For instance, in the illustrative example shown below, (i) the frequency band B<sub>2</sub> has one non-zero coefficient (represented by a line that extends above the lines 52 and 53) and four zero coefficients (represented by triangles on the x-axis), and (ii) the frequency band B<sub>3</sub> has three non-zero coefficients and two zero coefficients. In this case, Mahieux's system would simply determine that the frequency band B<sub>2</sub> has one non-zero coefficient, and separately determine that the frequency band B<sub>3</sub> has three non-zero coefficients (as shown on the left). This is

not the equivalent of determining the total number of the sub-bands  $B_1$ - $B_3$  that are audible (e.g., two total sub-bands,  $B_2$  and  $B_3$ ) (as shown in the right).

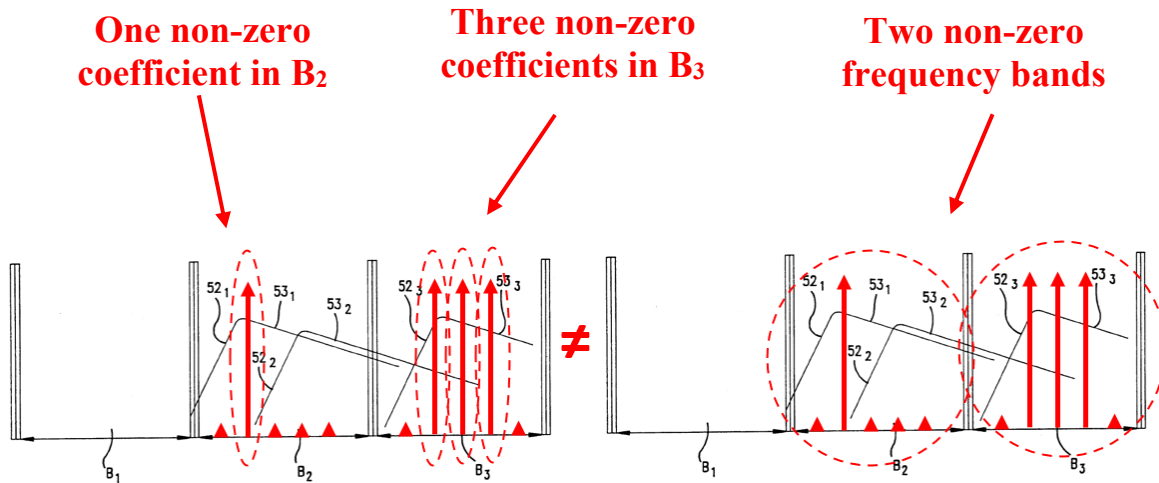


Exhibit 2003, FIG. 5 (revised and annotated)

82. As such, by making this comparison, Mahieux’s system does not determine “available *frequency-units*” at all (e.g., the “available” ones of frequency bands B<sub>1</sub>, B<sub>2</sub>, and B<sub>3</sub> alleged to be the “frequency-units”). Instead, at best, Mahieux’s system merely determines *spectral components* of the frequency-units that are individually “available” (e.g., the “available” ones of spectral lines 51<sub>1</sub>, 51<sub>2</sub>, and 51<sub>3</sub> within each of the frequency bands B<sub>1</sub>, B<sub>2</sub>, and B<sub>3</sub>). Indeed, as I explain in further detail below, this individual spectral component-based approach is critical to Mahieux’s system, as Mahieux’s system specifically relies upon the number of audible *spectral components* in each frequency band in order to select

from among several sets of quantizers for quantizing coefficients in that particular frequency band. Exhibit 2003, 11:39-62.

83. Further, even if Mahieux's system were to positively determine which frequency bands were audible (i.e., allegedly having one or more spectral components that exceed a threshold value), Mahieux's system still would not *count* the number of such audible frequency bands. For example, with respect to limitation [1D] of claim 1, the Petition appears to contend that the function " $\alpha(k)$ " represents the claimed "count," as it purportedly "accounts for 'the number of non-zero coefficients.'" Pet., 29 (first full paragraph). However, this function merely represents the number of constituent "non-zero *coefficients*" in a particular frequency band (e.g., representing respective *individual spectral components* of the frequency band), and not the number of "non-zero" *frequency bands* as a whole (e.g., among all of the frequency bands in the spectrum). Exhibit 2003, 12:22-29.

84. For instance, Mahieux describes that the value of the function " $\alpha(k)$ " is simply selected from a table of five pre-determined values, depending on the "number of non-zero coefficients." Exhibit 2003, 12:22-29 (explaining that the table contained five values that are "computed beforehand" and "is addressed by the number of non-zero coefficients"). Thus, at best, Mahieux merely describes determining a number of *spectral components* of one of Mahieux's "frequency-

units” (i.e., one of frequency band  $B_1$ ,  $B_2$ , and  $B_3$ ) that are individually “available,” rather than determining that a particular alleged “frequency-unit” (i.e., a particular band  $B_1$ ,  $B_2$ ,  $B_3$ ) as a whole is “available.”

85. Further, in connection with limitation [1D], the Petition appears to contend that the function “ $\beta(k)$ ” also represents the claimed “count,” as it purportedly “ensures that bits are allocated to ‘zones of the spectrum that contain peaks’ (that have non-zero coefficients).” Pet., (first full paragraph). However, “ $\beta(k)$ ” merely depends on a variable  $\gamma$  representing a “spectral spread” of a frequency band, and does not pertain at all to generating or otherwise using a *count* of a number of Mahieux’s audible frequency bands. Exhibit 2003, 12:29-56.

86. For instance, Mahieux describes that the variable  $\gamma$  is calculated according to the following equation:

$$\gamma = (1/N) \cdot \sum_{k=0}^{N-1} 10 \cdot \log_{10}(y^2(k)) - 10 \cdot \log_{10} \left( (1/N) \sum_{k=0}^{N-1} y^2(k) \right)$$

Exhibit 2003, 12:35-42.

87. Although Mahieux does not expressly describe each of the equation’s functions and/or parameters specifically in connection with this equation, elsewhere in its description, Mahieux describes that the function “ $y(k)$ ” represents the coefficients of the signal that are to be quantized, and the parameter “ $N$ ”

represents the number of samples per temporal block. *See, e.g.*, Exhibit 2003, 5:45-55, 5:62-64. Neither “ $y(k)$ ” nor “ $N$ ” is a count of a number of Mahieux’s audible frequency bands and, therefore, neither is a count of “available frequency units” as alleged in the Petition.

88. Further, a POSITA also would not have found it obvious to modify Mahieux’s system to count audible frequency bands (i.e., instead of audible spectral components of those frequency bands), as this count would not have served any purpose in Mahieux’s system. As I explained above, Mahieux’s system counts the number of audible *spectral components* in each frequency band in order to select from among several sets of quantizers for quantizing coefficients in that particular frequency band. Exhibit 2003, 11:39-62. For example, as shown in Mahieux’s FIG. 6, Mahieux’s system includes five sets of quantizers 61<sub>1</sub>-61<sub>5</sub>, each corresponding to a different number of “non-zero coefficients” (“ $N_{si}(j)$ ”) within a band (“ $j$ ”):

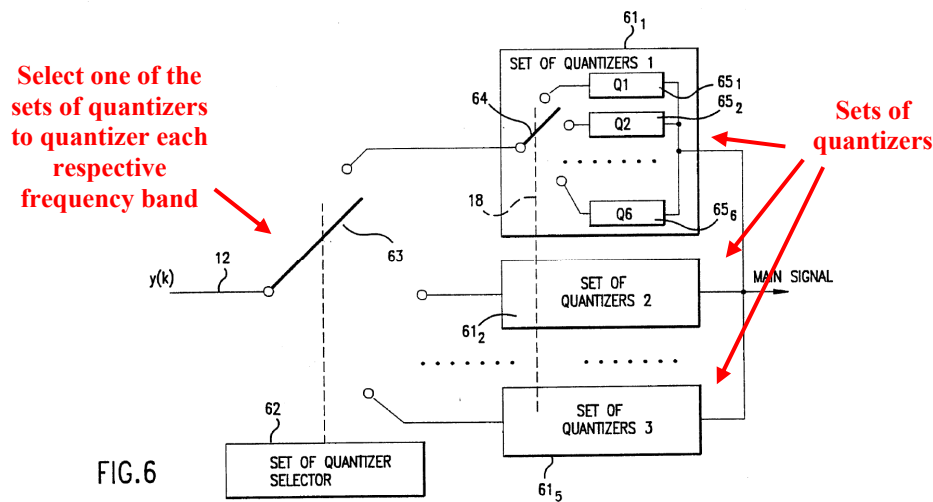


Exhibit 2003, FIG. 6 (annotated)

According to the invention, and so as to optimize the coding system, several sets of quantizers are available, as a function of  $N_{si}(j)$ . FIG. 6 shows the functional diagram of the module for selecting the quantizer to be used.

For example, in the following configuration, five sets  $61_1$  to  $61_5$  of quantizers are available. A module 62 for testing the value  $N_{si}(j)=1$  controls a selector 63 according to the following criteria:

- the first set for bands having one non-zero coefficient:  $N_{si}(j)=1$
- the second set for:  $N_{si}(j)=6$
- the third set for:  $6 > N_{si}(j) > 2$
- the fourth set for:  $10 > N_{si}(j) > 5$
- the fifth set for:  $N_{si}(j) > 9$ .

In each case, optimum quantizers for a number of bits varying, for example, between 1 and 6 bits, has been computed. A second selector 64 enables a choice to be made, in each set of quantizers  $65_1$  to  $65_6$ , of the optimal quantizer as a function of the number of bits  $R$  18.

Thus, the optimization of the choice of the quantizer is achieved as a function of the number of coefficients to be quantized and the number of bits allocated by the bit allocation module.

Exhibit 2003, 11:39-62.

89. That is, Mahieux specifically counts the number of non-zero coefficients in each frequency band in order to determine how to allocate bits *within that frequency band* (i.e., on a frequency-band-by-frequency-band basis). This makes intuitive sense, as a frequency band having a single non-zero coefficient can be quantized by allocating a relatively large number of bits to that single non-zero coefficient, whereas a frequency band having a large number of non-zero coefficients would require that fewer bits be allocated to each non-zero coefficient in order to ensure that each audible portion of the frequency band is represented. The function  $\alpha(k)$  accounts for this property (e.g., by selecting a performance factor corresponding to the number of non-zero coefficients).

90. For example, if a frequency band is allocated a particular number of bits for quantizing the coefficients within that frequency band, the system would need to determine how to distribute the bits among the coefficients. If the frequency band has only a single non-zero coefficient, the system can allocate substantially all of those bits to that single non-zero coefficient (e.g., as none of the other coefficients would need to be quantized and coded). However, if the frequency band has multiple non-zero coefficients, the system would allocate

fewer of those bits to each of the non-zero coefficient individually (e.g., such that each of the coefficients can be quantized and coded).

91. As an example, if the frequency band  $B_2$  were to have one non-zero coefficient, Mahieux's system would select the first set of quantizers to quantize coefficients in that frequency band. Exhibit 2003, 11:39-62 ("the first set [of quantizers] for bands having one non-zero coefficient:  $N_{si}(j)=1$ "). Further, if the frequency band  $B_3$  were to have three non-zero coefficients, Mahieux's system would select the third set of quantizers to quantize coefficients in that frequency band. Exhibit 2003, 11:39-62 ("the third set for:  $6 > N_{si}(j) > 2$ ").

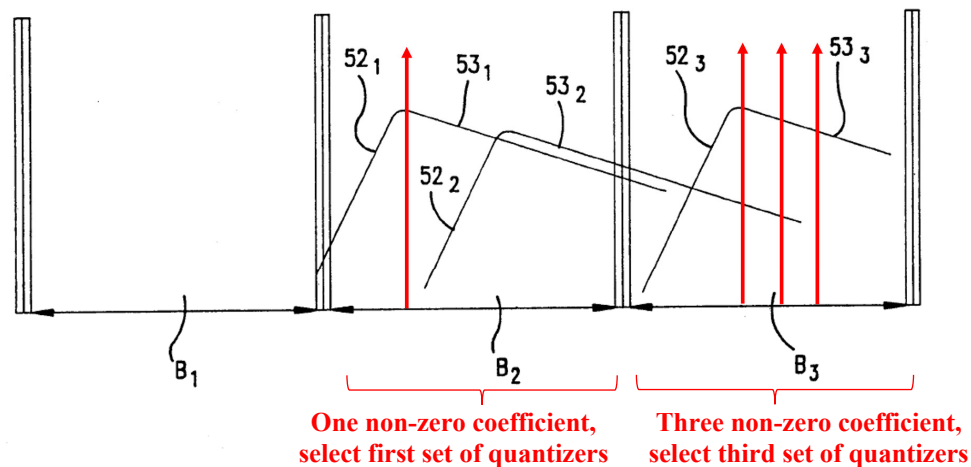


FIG.5

Exhibit 2003, FIG. 5 (revised and annotated)

92. It would not have made sense to modify Mahieux's system to instead count the number of audible frequency bands (e.g., the number of audible

frequency bands  $B_1$ - $B_3$ ), as such a count would play no role in Mahieux's bit allocation. Indeed, as Mahieux's bit allocation focuses on the number of non-zero coefficients in each individual frequency band in isolation and without regard to the aggregate number of total audible frequency bands of the overall spectrum, generating an aggregate number of total audible frequency bands (e.g., by counting) serves no purpose in Mahieux's allocation scheme. Further given Mahieux's focus on allocating bits to frequency coefficients in individual frequency bands in isolation from other bands, a POSITA would fail to find any use for such an aggregate number or count.

93. The Petition additionally refers to Fielder as purportedly "iterating through all frequency sub-bands to determine transform coefficients." Pet., 27 (first full paragraph) to 28 (first paragraph). However, even if Fielder's "iterating" process could be incorporated into Mahieux's system, at best, the Mahieux-Fielder system would simply "iterat[e] through *all* frequency sub-bands to determine transform coefficients," as alleged in Pet., 27 (first full paragraph, emphasis added). That is, the Mahieux-Fielder system still would not specifically count the number of Mahieux's "frequency bands" that are *audible* (i.e., the purported "available frequency-units").

94. Further, the count of the number of audible frequency bands does not play any role in Fielder's bit allocation. Instead, Fielder's bit allocation focuses on

assigning bits to (i) transform coefficients that have previously been assigned bits, and (ii) transform coefficients that are adjacent to those that have been previously assigned bits. Exhibit 2004, 23:13-29. *See also* Exhibit 2004, 23:30-25:13, FIG.

8. For example, as I explained in Section IX.B, Fielder's bit allocation process includes four specific "phases":

(i) "phase zero": "an initialization process for the remaining phases,"

(ii) "phase one": a phase in which the system "assigns bits, up to a maximum of four per transform coefficient, to the coefficients within the same critical band of those frequency components with the greatest spectral energy,

(iii) "phase two": a phase in which the system "allocates additional bits to the transform coefficients which were allocated bits during phase one such that the total adaptively allocated bits for each coefficient is four," and

(iv) "phase three": a phase in which the system "allocates bits to those coefficients which are adjacent to coefficients that were allocated bits during phase one and two." Exhibit 2004, 23:13-29.

95. None of these phases involves generating an aggregate number of total audible frequency bands (e.g., by counting). Thus, Fielder would not have

provided a POSITA with any motivation to utilize such information in Mahieux's system.

96. In sum, for at least the reasons discussed above, the Petition has failed to demonstrate that Mahieux discloses or renders obvious "a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria," either individually or in combination with Fielder.

**D. [Ground 1] Mahieux-Fielder Does Not Disclose "a First Bit Allocation Step for Determining an Initial Bit Allocation For Each of Said Available Frequency-Units By Using an Approximate Mathematical Model Using Said Counted Number of Available Frequency-Units," as Required by the Challenged Claims**

97. Claim 1 additionally recites "a first bit allocation step for determining an initial bit allocation for each of said available frequency-units by using an approximate mathematical model using said counted number of available frequency-units." The Petition alleges that Mahieux discloses and/or renders obvious this feature, relying on various operations described in Mahieux. Pet., Section VI.D.5. However, none of these operations amounts to *determining an initial bit allocation for each of said available frequency-units by using an approximate mathematical model using said counted number of available frequency-units*, as recited in claim 1.

98. As I explained in Section IX.C, Mahieux-Fielder does not disclose or render obvious "counting a number of available frequency-units," as recited in

claim 1. Accordingly, Mahieux-Fielder also does not disclose or render obvious using such a “count” at all, much less specifically “determining an initial bit allocation for each of said available frequency-units by using an approximate mathematical model *using said counted number of available frequency-units*,” as recited in claim 1.

99. In Section VI.D.5, the Petition alleges that Mahieux’s system allocates bits according to a ratio “ $\alpha(k) \cdot \beta(k) \cdot \sigma^2(k) / S(k)$ ,” and that such a ratio amounts to a “mathematical model.” Pet., 29 (first full paragraph) to 30 (first paragraph). However, to the extent that this ratio is a “mathematical model,” this ratio nevertheless does not use a count of Mahieux’s audible frequency bands (i.e., the alleged “available frequency-units”).

100. For example, Mahieux defines each of the functions and/or variables in the aforementioned ratio as follows:

(i)  $S(k)$ :

The allocation of the bits includes a module for computing the masking threshold  $S(k)$  (this computation is similar to the one performed for the detection of the masked lines) and a module 72 for quantizing the ratio  $\alpha(k) \cdot \beta(k) \cdot \sigma^2(k) / S(k)$  comparable to the one described by Yannick Mahieux in the article, "Transform Coding of Audio Signals Using Correlation Between Successive Transform Blocks" (Acts of the ICASSP Congress, Glasgow, 1989).

Exhibit 2003, 12:3-11.

(ii)  $\sigma^2(k)$ :

$\sigma^2(k)$  is the square of the spectrum descriptor, extended to all the coefficients  $y(k)$  of the band. It concerns the essential element of the ratio to be quantized. Since the coefficients are coded with different sets of quantizers (according to  $N_{si}(j)$ ), it is necessary to include the relative performance values of each set of quantizers in the allocation of the bits, according to the work by Jayant and Noll, *Digital Coding of Waveforms* (Prentice Hall Signal Processing Series, 1984).

Exhibit 2003, 12:12-21.

(iii)  $\alpha(k)$ :

The function  $\alpha(k)$  is equal to the performance factor of the quantizer to be used for the coefficient  $y(k)$ . A table 73 includes, for example, five values of  $\alpha(k)$  computed beforehand. This table is addressed by the number of non-zero coefficients. Taking account of the real performance values of the sets of quantizers enables a very appreciable improvement in the quality of the coding.

Exhibit 2003, 12:22-29.

(iv)  $\beta(k)$ :

The module 74 for computing the function  $\beta(k)$ , for its part, takes account of the spectral spread  $\gamma$ , also according to Jayant and Noll. This function indicates whether the spectrum is concentrated or not, and is computed at the coder by:

$$\gamma = (1/N) \cdot \sum_{k=0}^{N-1} 10 \cdot \log_{10}(y^2(k)) -$$
$$10 \cdot \log_{10} \left( (1/N) \sum_{k=0}^{N-1} y^2(k) \right)$$

For the allocation of the bits,  $\gamma$  is computed in four frequency bands of equal width. These four values are transmitted to the decoder by means of a uniform quantization on 6 bits.

In each of these four frequency bands, the function  $\beta(k)$  is computed according to the value of  $\gamma$  by means of a non-linear function. The role of  $\beta(k)$  is to force the allocation of the bits to grant a greater number of bits to the zones of the spectrum that contain peaks. Indeed, the coefficients  $y(k)$  corresponding to the pure sounds contained in the signal should be coded with higher precision, the masking threshold then having a level, as compared with that of the signal, that is lower than in the case of a noise spectrum.

Exhibit 2003, 12:47-56.

101. None of these functions and/or variables is or otherwise uses a count of a number of Mahieux's audible frequency bands (i.e., the alleged "available frequency-units"). Indeed, as I explained in Section IX.C, although the Petition specifically contends that the functions " $\alpha(k)$ " and/or " $\beta(k)$ " represents the claimed "count," neither of these functions represents the counted number of available frequency bands in Mahieux.

102. For instance, as I explained in Section IX.C, the function “ $\alpha(k)$ ” merely represents the number of constituent non-zero *coefficients* in a particular frequency band, and not the number of “non-zero” *frequency bands* as a whole. Exhibit 2003, 12:22-29. In particular, Mahieux describes that the value of the function “ $\alpha(k)$ ” is selected from a table of five pre-determined values, depending on the “number of non-zero coefficients.” Exhibit 2003, 12:22-29. As an example, if the number of non-zero coefficients is one, Mahieux’s system selects a value from the table corresponding to one non-zero coefficient. Further, if the number of non-zero coefficients is two, Mahieux’s system selects a value from the table corresponding to two non-zero coefficients, and so forth. Thus, at best, “ $\alpha(k)$ ” merely represents the number of *spectral components* that are individually “available” in each of the frequency-units, rather the number of frequency-units as a whole that are “available.”

103. Further, as I also explained in Section IX.C, the function “ $\beta(k)$ ” merely depends on a variable  $\gamma$  representing a “spectral spread” of a frequency band, and does not pertain at all to generating or otherwise using a *count* of a number of Mahieux’s audible frequency bands. For instance, as I explained in Section IX.C, Mahieux describes that the variable  $\gamma$  is calculated according to the following equation:

$$\gamma = (1/N) \cdot \sum_{k=0}^{N-1} 10 \cdot \log_{10}(y^2(k)) - 10 \cdot \log_{10} \left( (1/N) \sum_{k=0}^{N-1} y^2(k) \right)$$

Exhibit 2003, 12:35-42

104. Although Mahieux does not expressly describe each of the equation's functions and/or parameters specifically in connection with this equation, elsewhere in its description, Mahieux describes that the function "y(k)" represents the coefficients of the signal that are to be quantized, and the parameter "N" represents the number of samples per temporal block. *See, e.g.,* Exhibit 2003, 5:45-55, 5:62-64. Neither "y(k)" nor "N" is a count of a number of Mahieux's audible frequency bands.

105. Notably, the Petition does not assert that either of the functions  $\sigma^2(k)$  or S(k) represents the counted number of available frequency bands in Mahieux.

106. Moreover, as I also explained in Section IX.C, it would not have made sense to modify Mahieux's system to instead allocate based on the count of the number of audible frequency bands, as doing so would be contrary to Mahieux's technique of assigning bits to individual frequency bands in isolation and without regard to the aggregate number of total audible frequency bands of the overall spectrum. Indeed, Mahieux's system is configured to specifically count the number of non-zero coefficients in each frequency band—(on a frequency-by-

frequency-band basis) in order to determine how to allocate bits *within that frequency band*, such that different numbers of bits can be allocated to each frequency band depending on the characteristics of that frequency band in isolation. Exhibit 2003, 11:39-62. Allocating based on the count of the number of audible frequency bands would serve no purpose in Mahieux's system.

107. Fielder does not cure Mahieux's deficiencies. For example, as I explained in Section IX.C, although the Petition refers to Fielder as purportedly "iterating through all frequency sub-bands to determine transform coefficients," incorporating Fielder's process into Mahieux's system would, at best, result in a system that simply "iterat[es] through *all* frequency sub-bands to determine transform coefficients," as alleged by in Pet., 27 (first full paragraph, emphasis added). Indeed, as shown in Petitioner's annotation of Fielder's FIG. 7, Fielder's system simply repeats particular operations "for *all* subbands  $SB_i$ ," regardless of whether those subbands could be heard.

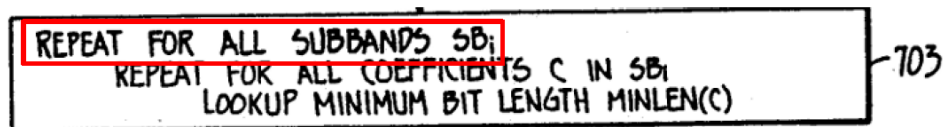


EXHIBIT 2004, FIG. 7 (as annotated in Pet., 28)

108. That is, the Mahieux-Fielder system still would not specifically count the number of Mahieux's "frequency bands" that are *audible* (*i.e.*, the purported "available frequency-units"), much less allocate based on such a count.

109. Further, the count of the number of audible frequency bands does not play any role in Fielder's bit allocation. Instead, Fielder's bit allocation focuses on assigning bits to (i) transform coefficients that have previously been assigned bits, and (ii) transform coefficients that are adjacent to those that have been previously assigned bits. Exhibit 2004, 23:13-29. *See also* Exhibit 2004, 23:30-25:13, FIG. 8.

110. For instance, as I explained in Sections IX.B-C, Fielder's bit allocation process includes four specific "phases":

- (i) "phase zero": "an initialization process for the remaining phases,"
- (ii) "phase one": a phase in which the system "assigns bits, up to a maximum of four per transform coefficient, to the coefficients within the same critical band of those frequency components with the greatest spectral energy,
- (iii) "phase two": a phase in which the system "allocates additional bits to the transform coefficients which were allocated bits during phase one such that the total adaptively allocated bits for each coefficient is four," and
- (iv) "phase three": a phase in which the system "allocates bits to those coefficients which are adjacent to coefficients that were allocated bits during phase one and two." Exhibit 2004, 23:13-29.

111. None of these phases involves generating an aggregate number of total audible frequency bands (e.g., by counting). Thus, Fielder would not have provided any motivation to utilize such information in Mahieux's system in order to allocate bits.

112. In sum, for at least the reasons discussed above, the Petition has failed to demonstrate that Mahieux discloses or renders obvious "a first bit allocation step for determining an initial bit allocation for each of said available frequency-units by using an approximate mathematical model using said counted number of available frequency-units," either individually or in combination with Fielder.

**E. Ground 2 Fails for the Same Reasons as for Ground 1**

113. The Petition contends that claims 1-17 are rendered obvious by Mahieux-Fielder, as modified by Akagari. Pet., Section VII.

114. With respect to claim 1, the Petition contends that Akagari provides additional disclosure regarding the claimed "representative." Pet., Section VII (e.g., Section VII.E, alleging that "[g]iven that Akagari discloses the representative in claims 4 and 5, it also discloses the representative in independent claim 1"). However, Akagari was not relied upon with respect to at least (i) "a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria" and (ii) "a first bit allocation step for determining an initial bit allocation for each of said available frequency-units by using an

approximate mathematical model using said counted number of available frequency-units,” as recited in claim 1.

115. Accordingly, Ground 2 also does not render obvious claim 1 or its respective dependent claims 1-17.

## **X. GROUNDS 3 AND 4**

116. The Petition alleges that (i) claims 1-3, 6-7, 10-11, 14, and 16 are obvious in view of Veldhuis and Galand (Ground 3), and (ii) claims 1-17 are obvious in view of Veldhuis, Galand, and Reaves (Ground 4). Pet., Sections VIII-IX.

117. However, as I explain below, these invalidity grounds fail to establish that the challenged claims are obvious for at least the following reasons.

### **A. Overview of Veldhuis**

118. In general, Veldhuis describes “[a] system for subband coding of a digital audio signal.” Exhibit 2006, Abstract.

119. During operation, the system receives a time-domain digital audio signal “ $x(k)$ ,” splits the signal into 26 different time-domain sub-band signals (e.g., using a filter bank), and quantizes the samples of each of the sub-band signals using an adaptive quantization technique. Exhibit 2006, 5:35-6:34.

120. For a first subset of the sub-band signals (i.e., sub-band signals 1-12), the system quantizes the samples by allocating bits according to a table (see FIG.

3) specifying a fixed quantization level for each of the sub-bands. Exhibit 2006, 9:44-66.

121. For a second subset of the sub-band signals (i.e., sub-band signals 13-26), the system quantizes the samples by initially allocating bits according to the same table (see FIG. 3). Exhibit 2006, 9:67-13:3. However, the system adaptively reallocates bits for some blocks in certain sub-bands, depending on whether a “characteristic parameter”  $G(p,m)$  for each block  $m$  in a certain sub-band  $p$  exceeds a threshold value  $T(p)$  (representing the range of human hearing for this sub-band  $p$ ). *Id.* The characteristics parameter  $G(p,m)$  is representative of the signal level in a block  $m$  and sub-band  $p$ . Exhibit 2006, 6:14-18.

122. As an example, if the characteristic parameter  $G(p,m)$  for a block  $m$  in the sub-band  $p$  is less than a threshold value  $T(p)$  (indicating that the samples within that block and sub-band cannot be heard), the system sets the bit allocation for the block  $m$  in the sub-band  $p$  to zero. Exhibit 2006, 11:14-49, 12:6-25. The bits that were previously allocated to the block  $m$  in the sub-band  $p$  are reallocated to other sub-bands (e.g., a block  $m$  in a different sub-band). Exhibit 2006, 12:25-59.

123. Conversely, if the characteristic parameter  $G(p,m)$  for a block  $m$  in the sub-band  $p$  exceeds the threshold value (indicating that the samples within that

block and sub-band can be heard), the system maintains the bit allocation for the block  $m$  in the sub-band  $p$ . Exhibit 2006, 11:67-12:7.

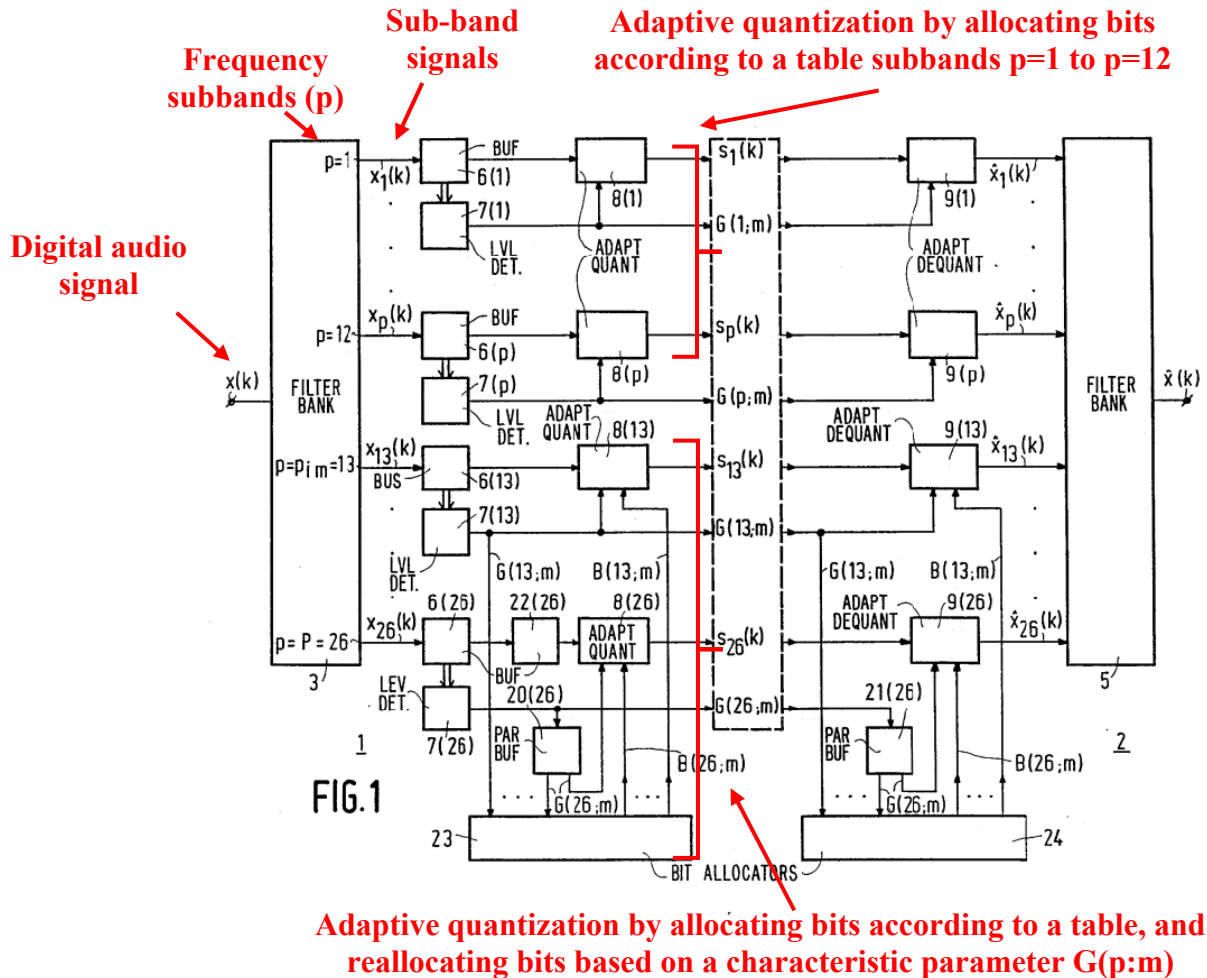
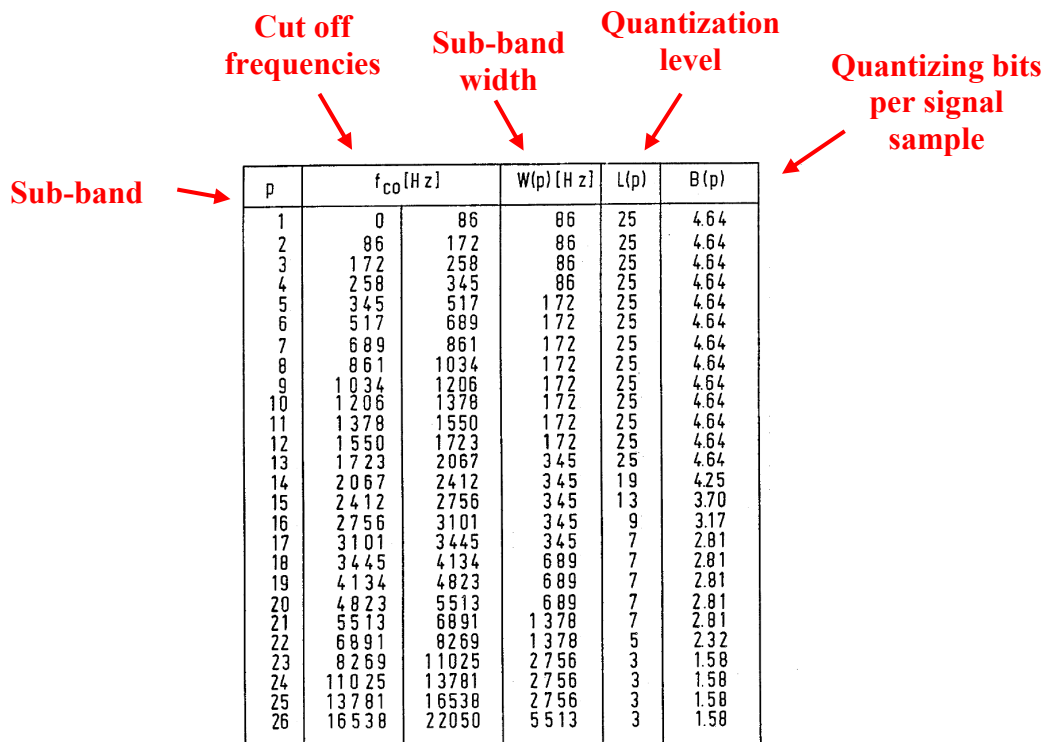


EXHIBIT 2004, FIG. 1 (annotated)



|  | p  | $f_{co}$ [Hz] | $W(p)$ [Hz] | $L(p)$ | $B(p)$ |
|--|----|---------------|-------------|--------|--------|
|  | 1  | 0             | 86          | 25     | 4.64   |
|  | 2  | 86            | 172         | 25     | 4.64   |
|  | 3  | 172           | 258         | 25     | 4.64   |
|  | 4  | 258           | 345         | 25     | 4.64   |
|  | 5  | 345           | 517         | 25     | 4.64   |
|  | 6  | 517           | 689         | 25     | 4.64   |
|  | 7  | 689           | 861         | 25     | 4.64   |
|  | 8  | 861           | 1034        | 25     | 4.64   |
|  | 9  | 1034          | 1206        | 25     | 4.64   |
|  | 10 | 1206          | 1378        | 25     | 4.64   |
|  | 11 | 1378          | 1550        | 25     | 4.64   |
|  | 12 | 1550          | 1723        | 25     | 4.64   |
|  | 13 | 1723          | 2067        | 25     | 4.64   |
|  | 14 | 2067          | 2412        | 19     | 4.25   |
|  | 15 | 2412          | 2756        | 13     | 3.70   |
|  | 16 | 2756          | 3101        | 9      | 3.17   |
|  | 17 | 3101          | 3445        | 7      | 2.81   |
|  | 18 | 3445          | 4134        | 7      | 2.81   |
|  | 19 | 4134          | 4823        | 7      | 2.81   |
|  | 20 | 4823          | 5513        | 7      | 2.81   |
|  | 21 | 5513          | 6891        | 7      | 2.81   |
|  | 22 | 6891          | 8269        | 5      | 2.32   |
|  | 23 | 8269          | 11025       | 3      | 1.58   |
|  | 24 | 11025         | 13781       | 3      | 1.58   |
|  | 25 | 13781         | 16538       | 3      | 1.58   |
|  | 26 | 16538         | 22050       | 3      | 1.58   |

EXHIBIT 2004, FIG. 3 (annotated)

## B. Overview of Galand

124. In general, Galand describes “a[] voice signal coding method ... for processing the residual base-band signal provided by a Voice Excited Predictive (VEPC) Coder.” Exhibit 2007, Abstract.

125. In one aspect, Galand’s system receives a time-domain digital audio signal, splits the signal into eight different time-domain sub-band signals (e.g., using a filter bank), and quantizes the samples of a subset of the sub-band signals (i.e., sub-bands 2-7) using a “dynamic bit allocating device” (DAB). Exhibit 2007, 4:33-6:2.

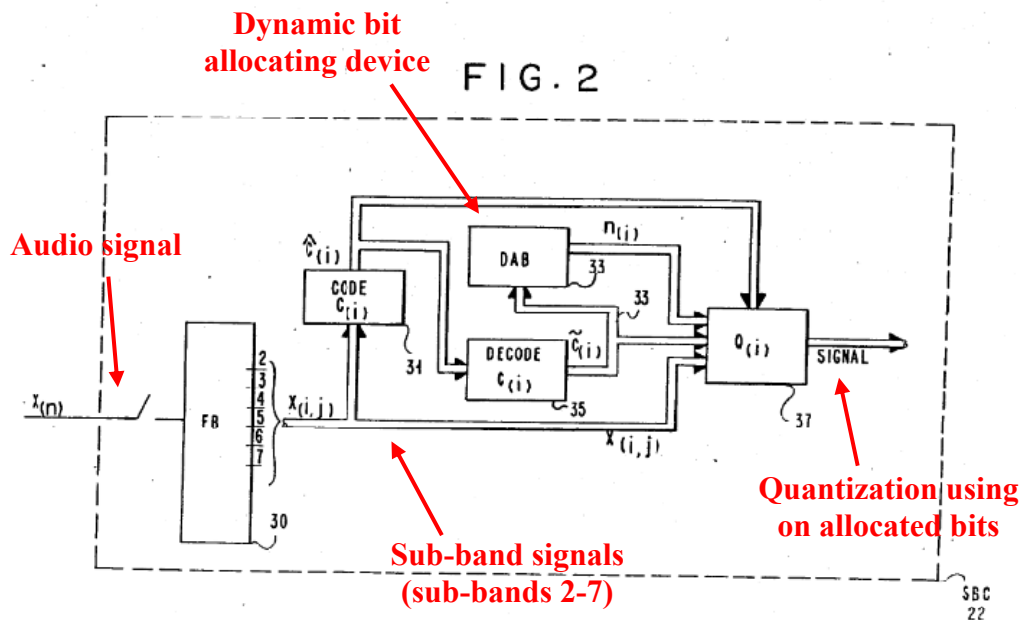


EXHIBIT 2007, FIG. 2 (annotated)

126. Galand's system allocates bits for quantization using the following equation:

$$n(i) = \frac{1}{P} \left[ M - \sum_{j=2}^7 \log_2 \tilde{C}(j) \right] + \log_2 \tilde{C}(i)$$

Exhibit 2007, 5:1.

127. According to this equation:

- $n(i)$  represents "the bit rate ... to be allocated for quantizing the  $i^{\text{th}}$  sub-band samples,"
- $M$  represents "the total number of bits to be distributed over the  $p = 6$  sub-bands,"
- $P$  is equal to six (i.e., a fixed number of sub-bands), and

- $C(i)$  is the maximum sample in sub-band  $i$ , among samples  $X(i,j)$

(“transcoded” into  $\tilde{C}(i)$ ). Exhibit 2007, 4:33-5:7.

**C. [Ground 3] Veldhuis-Galand Does Not Disclose “A Step For Counting A Number Of Available Frequency-Units By Using The Representative, Based On Psychoacoustic Criteria,” As Required By The Challenged Claims**

128. Claim 1 recites “a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria.” The Petition alleges that Veldhuis discloses and/or renders obvious this feature, relying on various operations described in Veldhuis. Pet., Section VIII.D.4. However, none of these operations amounts to *counting a number of available frequency-units*, as recited in claim 1.

129. In Section VIII.2, the Petition alleges that Veldhuis describes a coder that “divides the ... signal band of 0-22.05 kHz” “into a number of  $P=26$  subbands,” and “further samples each subband in ‘a time interval corresponding with one block of  $M=32$ .’” Pet., 55 (second full paragraph), quoting EX1008, 5:53-59, 9:67-10:13. In particular, the Petition contends that “[e]ach of these *blocks* ( $p;m$ ) represents the claimed ‘frequency-unit.’” Pet., 55 (second full paragraph, emphasis added). Each of Veldhuis’s “blocks” is a sub-band signal (i.e., time-domain signals) that is obtained by filtering the audio signal by a filter bank. Exhibit 2006, 5:53-68.

130. Further, the Petition alleges that Veldhuis's blocks are "available" if they "can be heard and thus are available for bit allocation" (e.g., if a characteristic parameter  $G(p;m)$  of the block is greater than a threshold  $T(p)$ , indicating that the block has a "power" that is greater than a "masking threshold corresponding to the human auditory system"). Pet., 57 (fourth full paragraph) to 58 (second full paragraph). The Petition also contends that "[b]y determining which signal blocks can be heard and thus are available for bit allocation (those having  $C(p;m)$  values equal to 1), Veldhuis "counts" such blocks." Pet. 58 (second full paragraph)

131. However, even under the Petition's construction, Veldhuis fails to describe **counting** the number of Veldhuis's "blocks" that can be heard (i.e., the purported "available frequency-units").

132. As an initial matter, the Petition notes that during prosecution of the '757 patent, the Board of Patent Appeals and Interference (BPAI) found that Veldhuis '463 failed to disclose "counting" the "available frequency-units." Exhibit 2012, 259. Veldhuis '463 is a divisional application of Veldhuis with substantively the same disclosure as Veldhuis. *See generally* Exhibit 2009. However, the Petition contends that "nothing in the claims requires actively counting by, for example, incrementing a counter." Pet., 58 (third full paragraph) to 59 (first paragraph).

133. I disagree with the Petition's characterization of the claim. Although the claims do not expressly require actively counting by incrementing a bandwidth counter (e.g., as described in the specification of the '757 patent), claim 1 nevertheless positively recites a "step for counting a number of available frequency-units." Further, as I explain in Section X.D below with respect to limitation [1D] of claim 1, the claims expressly recite that this "counted number" is used to perform additional operations, namely performing "a first bit allocation step for determining an initial bit allocation for each of said available frequency-units." That is, "counting" is a positive limitation that is integral to claimed subject matter, and cannot be ignored.

134. Further, I disagree with the Petition's allegation that "the BPAI failed to consider Veldhuis's ['463] entire disclosure" (and by virtue, Veldhuis's disclosure) in finding that Veldhuis '463 failed to disclose performing such a "count," as the Petition has misrepresented key aspects of the disclosure in Veldhuis '463 (and correspondingly the disclosure in Veldhuis).

135. By way of background, Veldhuis describes a system that (i) receives a time-domain digital audio signal, (ii) splits the signal into 26 different time-domain sub-band signals (e.g., using a filter bank), and (iii) quantizes the samples of each of the sub-band signals using a block adaptive pulse code modulation (PCM) method and adaptive quantization technique. Exhibit 2006, 5:53-6:9.

136. Veldhuis's system quantizes samples differently depending on the sub-band of the sample. Exhibit 2006, 9:44-11:13. For example, for a first subset of the sub-band signals (i.e., sub-band signals 1-12), Veldhuis's system quantizes the samples by allocating bits according to a table specifying a fixed quantization level for each of the sub-bands and a corresponding number of bits. Exhibit 2006, 9:60-66. This table is shown in FIG. 3.

|    | <b>Sub-band</b> | <b>Cut off frequencies</b> | <b>Sub-band width</b> | <b>Quantization level</b> | <b>Quantizing bits per signal sample</b> |        |
|----|-----------------|----------------------------|-----------------------|---------------------------|------------------------------------------|--------|
|    | $p$             | $f_{co} [Hz]$              |                       | $W(p) [Hz]$               | $L(p)$                                   | $B(p)$ |
| 1  | 0               | 86                         | 86                    | 25                        | 4.64                                     |        |
| 2  | 86              | 172                        | 86                    | 25                        | 4.64                                     |        |
| 3  | 172             | 258                        | 86                    | 25                        | 4.64                                     |        |
| 4  | 258             | 345                        | 86                    | 25                        | 4.64                                     |        |
| 5  | 345             | 517                        | 172                   | 25                        | 4.64                                     |        |
| 6  | 517             | 689                        | 172                   | 25                        | 4.64                                     |        |
| 7  | 689             | 861                        | 172                   | 25                        | 4.64                                     |        |
| 8  | 861             | 1034                       | 172                   | 25                        | 4.64                                     |        |
| 9  | 1034            | 1206                       | 172                   | 25                        | 4.64                                     |        |
| 10 | 1206            | 1378                       | 172                   | 25                        | 4.64                                     |        |
| 11 | 1378            | 1550                       | 172                   | 25                        | 4.64                                     |        |
| 12 | 1550            | 1723                       | 172                   | 25                        | 4.64                                     |        |
| 13 | 1723            | 2067                       | 345                   | 25                        | 4.64                                     |        |
| 14 | 2067            | 2412                       | 345                   | 19                        | 4.25                                     |        |
| 15 | 2412            | 2756                       | 345                   | 13                        | 3.70                                     |        |
| 16 | 2756            | 3101                       | 345                   | 9                         | 3.17                                     |        |
| 17 | 3101            | 3445                       | 345                   | 7                         | 2.81                                     |        |
| 18 | 3445            | 4134                       | 689                   | 7                         | 2.81                                     |        |
| 19 | 4134            | 4823                       | 689                   | 7                         | 2.81                                     |        |
| 20 | 4823            | 5513                       | 689                   | 7                         | 2.81                                     |        |
| 21 | 5513            | 6891                       | 1378                  | 7                         | 2.81                                     |        |
| 22 | 6891            | 8269                       | 1378                  | 5                         | 2.32                                     |        |
| 23 | 8269            | 11025                      | 2756                  | 3                         | 1.58                                     |        |
| 24 | 11025           | 13781                      | 2756                  | 3                         | 1.58                                     |        |
| 25 | 13781           | 16538                      | 2756                  | 3                         | 1.58                                     |        |
| 26 | 16538           | 22050                      | 5513                  | 3                         | 1.58                                     |        |

EXHIBIT 2004, FIG. 3 (annotated)

137. For a second subset of the sub-band signals (i.e., sub-band signals 13-26), Veldhuis's system quantizes the samples by initially allocating bits according to the same table (*see* FIG. 3). Exhibit 2006, 9:67-10:14; 10:31-35. However, the

system adaptively reallocates bits for some blocks in certain sub-bands, depending on whether a “characteristic parameter”  $G(p,m)$  for each block  $m$  in a certain sub-band  $p$  (indicating the signal level of the block in the sub-band) exceeds a threshold value  $T(p)$  representing the range of human hearing for this sub-band  $p$ . Exhibit 2006, 9:67-12:59.

138. As an example, when the characteristic parameter  $G(p,m)$  for a block  $m$  in the sub-band  $p$  is *less* than a threshold value  $T(p)$ , this indicates that the samples within that block and sub-band *cannot be heard*. Exhibit 2006, 11:14-49, 12:6-25. In this case, the system *zeros out* the bit allocation for the block  $m$  in the sub-band  $p$  and adds the bits to a sum of bits “S” (representing the number of saved bits for reallocation). *Id.* This is also reflected in FIG. 8 (operations 8-5 and 8-6), whereby when a binary comparator signal “C” for a block “m” in subband “p” is zero (indicating that the block is less than a threshold value and cannot be heard), the number of bits “B” allocated to that block is set to zero. The sum of bits “S” are subsequently reallocated to other sub-bands (e.g., a block  $m$  in a different sub-band). Exhibit 2006, 12:25-59.

139. Conversely, when the characteristic parameter  $G(p,m)$  for a block  $m$  in the sub-band  $p$  *exceeds* the threshold value, this indicates that the samples within that block and sub-band *can be heard*. Exhibit 2006, 11:67-12:7. In this case, the system maintains the bit allocation for the block  $m$  in the sub-band  $p$ . *Id.*

This is also reflected in FIG. 8 (operation 8-5), whereby when a binary comparator signal “C” for a block “m” in subband “p” is not zero (indicating that the block is greater than or equal to a threshold value and can be heard), the number of bits “B” allocated to that block is not changed.

140. That is, Veldhuis selectively increments a “*sum of saved bits S*,” which is not the number of blocks that can be heard by a human (i.e., the alleged “available frequency-units”). Instead, as shown in column B(p) of the table in FIG. 3, the number of bits that are allocated to each block (and potentially reallocated to other blocks) are different respective decimal values that obfuscate the number of blocks that can or cannot be heard. Further, the number of bits in the table are pre-determined, and do not depend on the number of blocks that are audible.

141. Further, Veldhuis increments the “sum of saved bits S” only when blocks *cannot be heard* by a human, in order to determine how many bits that were to be allocated to that block can be reallocated to other blocks instead (i.e., blocks that can be heard). Thus, even if “the sum of saved bits S” were somehow alleged to inform a “count” of “frequency-units” (which Patent Owner does not concede), at best Veldhuis would describe the *very opposite* of that required by claim (i.e., counting a number of *unavailable* frequency-units, rather than available

frequency-units) because the sum of “S” depends from unavailable/inaudible blocks, rather than available/audible blocks.

142. Further, it would not have been obvious to modify Veldhuis’s system to count the number of blocks that can be heard by a human instead (i.e., the alleged “available frequency-units”) because such a parameter would not serve any purpose in Veldhuis’s system. In particular, Veldhuis’s system does not reallocate bits from any blocks that can be heard. Exhibit 2006, 11:67-12:7. Instead, Veldhuis’s maintains the number of allocated bits for those blocks, and in some cases selectively allocates one or more additional bits to those blocks from blocks that cannot be heard. *Id.* As Veldhuis does not reallocate any bits from audible blocks and merely allocates one or more additional bits to the audible blocks depending on the number of inaudible blocks, the specific number of *audible* blocks that exist is irrelevant to Veldhuis’s system. As such, in Veldhuis’s system, tracking the number of blocks that are audible to a human would be entirely superfluous.

143. Alternatively, the Petition alleges that Galand also describes “counting” available frequency-units, and cites to a portion in Galand describing “useful sub-bands.” Pet., 60 (second full paragraph) to 61 (first full paragraph). However, for at least the following reasons, Galand’s “useful” sub-bands are not analogous to Veldhuis’s audible blocks (i.e., the alleged “available frequency-

units”) and thus would not have motivated a POSITA to modify Veldhuis’s system to count such blocks.

144. As an initial matter, Galand does not expressly describe what a “useful sub-band” entails. However, Galand refers to dividing a frequency range into eight sub-bands, and considering a fixed number of sub-bands 2-7 only. *See, e.g.,* Exhibit 2007, 4:33-38 (emphasis added):

FIG. 2 schematically shows sub-band coder 22 which is provided with filter set (FB) 30 which receives samples  $X_n$ . This filter set divides the pass band of the residual base-band signal into eight adjacent sub-bands 1-8. Only sub-bands 2 to 7 are kept for the needs of the invention.

*See also* Exhibit 2007, FIG. 2 (depicting that the output of the filter set “FB” only includes sub-bands 2-7). Further, Galand does not expressly describe that “useful” sub-bands are sub-bands that can be heard, and that any remaining sub-bands cannot be heard (as the Petition appears to imply). Thus, Galand’s “useful” sub-bands are not analogous to Veldhuis’s audible “blocks” and are not interchangeable with one another.

145. Further, even if this distinction were to be ignored, Galand does not describe counting “useful” sub-bands “by using the representative,” as required by [1C]. For instance, in Galand, the “useful sub-bands” appears to refer to a ***pre-determined*** set of sub-bands (i.e., sub-bands 2-7) that will be processed by the

system. Exhibit 2007, 4:33-38. As these sub-bands are predetermined in some undisclosed way, Galand fails to disclose that these sub-bands are determined, identified, or counted by using a “representative” of each of the sub-bands, much less using the “characteristic parameter  $G(p;m)$ ” or equivalent thereof that was identified by the Petition in its mapping. As such, even if Galand were to describe counting “available frequency-units,” Galand still would not teach doing so based specifically on a representative of those frequency-units, as recited in claim 1.

146. In sum, for at least the reasons discussed above, the Petition has failed to demonstrate that Veldhuis discloses or renders obvious “a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria,” either individually or in combination with Galand.

**D. [Ground 3] Veldhuis-Galand Does Not Disclose “a First Bit Allocation Step for Determining an Initial Bit Allocation For Each of Said Available Frequency-Units By Using an Approximate Mathematical Model Using Said Counted Number of Available Frequency-Units,” as Required by the Challenged Claims**

147. Claim 1 additionally recites “a first bit allocation step for determining an initial bit allocation for each of said available frequency-units by using an approximate mathematical model using said counted number of available frequency-units.” The Petition alleges that Veldhuis discloses and/or renders obvious this feature, relying on various operations described in Veldhuis. Pet., Section VIII.D.5. However, none of these operations amounts to *determining an*

*initial bit allocation for each of said available frequency-units by using an approximate mathematical model using said counted number of available frequency-units*, as recited in claim 1.

148. As I explained in Section X.C, Veldhuis-Galand does not disclose or render obvious “counting a number of available frequency-units,” as recited in claim 1. Accordingly, Veldhuis-Galand also does not disclose or render obvious using such a “count” at all, much less specifically “determining an initial bit allocation for each of said available frequency-units by using an approximate mathematical model *using said counted number of available frequency-units*,” as recited in claim 1. Indeed, the Petition contends that Veldhuis describes the claimed “first bit-allocation step,” as Veldhuis describes a process in which “bits  $B(p)$  are allocated to blocks that can be heard (those having  $C(p;m)=1$ ), while no bits are allocated to blocks that cannot be heard (those having  $C(p;m)=0$ ).” Pet, 61 (third full paragraph). However, even if this operation in Veldhuis were to be considered a “first bit allocation step,” as the Petition appears to contend, Veldhuis’s operation is not performed “using an approximate mathematical model.” Pet. 62, (first full paragraph, conceding that “Veldhuis does not expressly disclose how the  $L(p)$  values (and thus the  $B(p)$  values) are determined”).

149. To remedy this deficiency, the Petition relies on Galand's teachings, contending that the Veldhuis-Galand system would allocate bits according to the following equation described in Galand:

$$n_{(i)} = \frac{1}{p} \left[ M - \sum_{j=2}^7 \log_2 \tilde{C}_{(j)} \right] + \log_2 \tilde{C}_{(i)}$$

Exhibit 2007, 5:1-5. In doing so, the Petition appears to contend that the value "p = 6" amounts to the claimed "counted number of available frequency-units," and as such the equation represents "an approximate mathematical model using said counted number of available frequency-units." Pet., 62 (first full paragraph) to 63 (first full paragraph).

150. However, the Petition's characterization of Galand is inconsistent with Galand's disclosure. As I explained in Section X.C, claim 1 requires that the "counted number of available frequency-units" (e.g., that are used to "determine[e] an initial bit allocation for each of said available frequency-units") is determined "by using the representative" of each frequency-unit. In contrast, Galand describes dividing a frequency range into eight sub-bands, and considering a *fixed number* of sub-bands 2-7 only, resulting in a total of **6 subbands** (i.e., corresponding to "p = 6" in the equation above). Exhibit 2007, 4:33-38. That is, these sub-bands are a *pre-determined* set of sub-bands (i.e., sub-bands 2-7) that will be processed by the

system. As these sub-bands are predetermined in some undisclosed way, Galand fails to disclose that these sub-bands are determined, identified, or counted by using a “representative” of each of the sub-bands, as required by claim 1. Further, there is no indication in Galand that these sub-bands are specifically those that can be heard by a human (i.e., the alleged criteria for determining whether a frequency-unit is “available”). Indeed, Galand’s system also selects the same six sub-bands by default, seemingly without considering the actual contents of the specific audio signals—and their sub-band signals—that are input into the system.

151. Further, even if Galand’s deficiencies were to be ignored, there nevertheless would not have been any expectation of success in using Galand’s equation in Veldhuis’s system in the manner set forth by the Petition, as such a modification would be contrary to Veldhuis’s multi-step dynamic bit allocation process.

152. As I explained above, Veldhuis’s system quantizes the samples by initially allocating bits according to a table, an example of which is shown in FIG. 3. Exhibit 2006, 9:67-10:14, 10:31-3, FIG. 3. However, the system subsequently adaptively reallocates bits for some blocks in certain sub-bands, depending on whether the block can be heard. That is, Veldhuis’s system requires that bits be *initially allocated to each of the blocks* (e.g., regardless of whether they can be

heard or not), such that some of those bits (e.g., those allocated to inaudible blocks) can be *subsequently reallocated* to other blocks.

153. This multi-step bit allocation process is reflected in the table shown in Veldhuis's FIG. 3, which indicates the initial allocation of bits to each and every sub-band, regardless of whether the blocks in that sub-band can be heard.

|  | p  | f <sub>co</sub> [Hz] |       | W(p) [Hz] | L(p) | B(p) |
|--|----|----------------------|-------|-----------|------|------|
|  | 1  | 0                    | 86    | 86        | 25   | 4.64 |
|  | 2  | 86                   | 172   | 86        | 25   | 4.64 |
|  | 3  | 172                  | 258   | 86        | 25   | 4.64 |
|  | 4  | 258                  | 345   | 86        | 25   | 4.64 |
|  | 5  | 345                  | 517   | 172       | 25   | 4.64 |
|  | 6  | 517                  | 689   | 172       | 25   | 4.64 |
|  | 7  | 689                  | 861   | 172       | 25   | 4.64 |
|  | 8  | 861                  | 1034  | 172       | 25   | 4.64 |
|  | 9  | 1034                 | 1206  | 172       | 25   | 4.64 |
|  | 10 | 1206                 | 1378  | 172       | 25   | 4.64 |
|  | 11 | 1378                 | 1550  | 172       | 25   | 4.64 |
|  | 12 | 1550                 | 1723  | 172       | 25   | 4.64 |
|  | 13 | 1723                 | 2067  | 345       | 25   | 4.64 |
|  | 14 | 2067                 | 2412  | 345       | 19   | 4.25 |
|  | 15 | 2412                 | 2756  | 345       | 13   | 3.70 |
|  | 16 | 2756                 | 3101  | 345       | 9    | 3.17 |
|  | 17 | 3101                 | 3445  | 345       | 7    | 2.81 |
|  | 18 | 3445                 | 4134  | 689       | 7    | 2.81 |
|  | 19 | 4134                 | 4823  | 689       | 7    | 2.81 |
|  | 20 | 4823                 | 5513  | 689       | 7    | 2.81 |
|  | 21 | 5513                 | 6891  | 1378      | 7    | 2.81 |
|  | 22 | 6891                 | 8269  | 1378      | 5    | 2.32 |
|  | 23 | 8269                 | 11025 | 2756      | 3    | 1.58 |
|  | 24 | 11025                | 13781 | 2756      | 3    | 1.58 |
|  | 25 | 13781                | 16538 | 2756      | 3    | 1.58 |
|  | 26 | 16538                | 22050 | 5513      | 3    | 1.58 |

EXHIBIT 2004, FIG. 3 (annotated)

154. In contrast, Galand's equation is used to determine how many bits M are distributed among six pre-determined sub-bands (i.e., p = 6) in a single allocation step. Exhibit 2007, 4:64-7. If Galand's equation were to be modified to replace the pre-determined value of "p" with the number of Veldhuis's block that

can be heard (i.e., a dynamically changing value, depending on the contents of the audio signal), at best, the equation would determine how many bits  $M$  are distributed among *audible blocks only*. That is, all of Veldhuis's bits would be allocated in a *single step* (i.e., to the audible blocks only in accordance with Galand's modified equation), and there would be no remaining bits to reallocate in Veldhuis's subsequent reallocation step. Thus, Veldhuis's subsequent reallocation step, which Petitioner relies on for the recited "second bit allocation step," would be rendered entirely superfluous.

155. Further, making such a substitution also would be contrary to Galand's teaching. In particular, this substitution would prevent bits from being distributed across the fixed number of frequency sub-bands that are retained by a filter bank (i.e., the very purpose of the Galand's equation), as that fixed number would no longer be represented in the equation at all. For example, in Galand's equation, the term " $1/p$ " is used to distribute bits across a fixed number of sub-bands (i.e., by allocating to each sub-band " $i$ " one-sixth of a particular number of bits, as adjusted by a term " $\log_2 \hat{C}(i)$ "). By substituting the term " $1/p$ " with another term, the equation would no longer account for the specific six fixed sub-bands identified as Galand as being "useful," and thus would fail to effectively distribute bits among them as taught by Galand.

156. In sum, for at least the reasons discussed above, the Petition has failed to demonstrate that Veldhuis-Galand discloses or renders obvious “a first bit allocation step for determining an initial bit allocation for each of said available frequency-units by using an approximate mathematical model using said counted number of available frequency-units.”

**E. Ground 4 Fails for the Same Reasons as for Ground 3**

157. The Petition contends that claims 1-17 are rendered obvious by Veldhuis-Galand, as modified by Reaves. Pet., Section IX.

158. With respect to claim 1, the Petition contends that Reaves provides additional disclosure regarding the claimed “representative.” Pet., Section IX (e.g., Section IX.E, alleging that “given that Reaves discloses the representative in claims 4-5, it also discloses the representative in independent claim 1”). However, Reaves was not relied upon with respect to at least (i) “a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria” and (ii) “a first bit allocation step for determining an initial bit allocation for each of said available frequency-units by using an approximate mathematical model using said counted number of available frequency-units,” as recited in claim 1.

159. Accordingly, Ground 4 also does not render obvious claim 1 or its respective dependent claims 1-17.

## **XI. CONCLUSION**

160. The findings and opinions set forth in this declaration are based on my work and examinations to date.

161. I may continue my examinations. I may also receive additional documentation and other factual evidence over the course of this IPR that will allow me to supplement and/or refine my opinions. I reserve the right to add to, alter, or delete my opinions and my declaration upon discovery of any additional information. I reserve the right to make such changes as may be deemed necessary.

162. In signing this declaration, I recognize that the declaration will be filed as evidence in an IPR before the PTAB. I also recognize that I may be subject to cross-examination in the case and that cross-examination will take place within the United States. If cross-examination is required of me, I will appear for cross-examination within the United States during the time allotted for cross-examination.

163. I hereby 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 further that these statements were made with the knowledge that willful false statements and the like are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code.

Declaration of Chris Kyriakakis  
Exhibit 2002

Dated: 01/08/2025

By:   
Chris Kyriakakis

# EXHIBIT A

## Prof. Chris Kyriakakis

### Contact

USC Viterbi School of Engineering (1996 – Present)  
Ming Hsieh Department of Electrical Engineering  
email: [ckyriak@me.com](mailto:ckyriak@me.com)  
Wikipedia: [http://en.wikipedia.org/wiki/Chris\\_Kyriakakis](http://en.wikipedia.org/wiki/Chris_Kyriakakis)

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### Education

|                    |                                                                                        |
|--------------------|----------------------------------------------------------------------------------------|
| <b>PhD</b><br>1993 | <b>University of Southern California (USC)</b><br>Electrical Engineering               |
| <b>MS</b><br>1987  | <b>University of Southern California (USC)</b><br>Electrical Engineering               |
| <b>BS</b><br>1985  | <b>California Institute of Technology (Caltech)</b><br>Engineering and Applied Science |

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### Positions

|                                              |                                      |
|----------------------------------------------|--------------------------------------|
| <b>Associate Professor</b><br>2002-Present   | <b>USC</b><br>Electrical Engineering |
| <b>Assistant Professor</b><br>1996-2002      | <b>USC</b><br>Electrical Engineering |
| <b>Research Fellow</b><br>1993-1996          | <b>USC</b><br>Electrical Engineering |
| <b>CTO</b><br>2002-2018                      | <b>Audyssey Laboratories</b>         |
| <b>Chief Audio Scientist</b><br>2019-Present | <b>SYNG</b>                          |

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### Research Areas

Immersive audio signal processing, machine learning for audio and acoustics, room equalization, room acoustics, headphone acoustics, multichannel audio streaming, microphone array processing, head-related transfer function modeling, adaptive algorithms for virtual microphones, virtual loudspeakers, multichannel audio rendering, archaeoacoustics.

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### Honors and Awards

|      |                                                                                          |
|------|------------------------------------------------------------------------------------------|
| 2018 | Elected as Senior Member of the Institute of Electrical and Electronics Engineers (IEEE) |
| 2006 | World Technology Network Award                                                           |

James P. Clark, Chairman of the World Technology Network: “The World Technology Awards program is a very inspiring way to identify and honor the most innovative people and organizations in the technology world...We look forward to assisting Prof. Kyriakakis continue to help create our collective future and change our world.”

- 2004 National Academy of Engineering: Frontiers of Engineering Session Chairman
- 2003 Best Paper Award: IEEE 37<sup>th</sup> Asilomar Conference on Signals, Systems, & Computers

## **Professional Service (Past and Current)**

### **Society Memberships**

Member of the Audio Engineering Society (AES)  
 Senior Member of the IEEE  
 Member of the Acoustical Society of America (ASA)  
 Member IEEE Technical Committee on Multimedia  
 Member of AES Technical Committee on Multichannel and Binaural Audio  
 Member of AES Technical Committee on Audio for Games

### **Reviewer**

National Science Foundation  
 IEEE Transactions on Multimedia  
 IEEE Transactions on Signal Processing  
 IEEE Transactions on Audio, Speech, and Language Processing  
 European Association for Signal Processing (EURASIP)  
 IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP)  
 IEEE Workshop on Applications of Signal Processing to Audio (WASPAA)  
 IEEE Workshop on Multimedia Signal Processing  
 ACM Special Interest Group on Graphics and Interactive Techniques (SIGGRAPH)  
 European Signal Processing Conference (EUSIPCO)  
 Audio Engineering Society  
 Acoustical Society of America

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## Books

1. S. Bharitkar and C. Kyriakakis, “Immersive Audio Signal Processing,” Springer-Verlag, New York, (2006).
2. D. Yang, C.-C. Kuo, and C. Kyriakakis, “High Fidelity Multichannel Audio Coding,” Hindawi Press, New York, (2004).

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## Patents

1. Systems and Methods for Audio Upmixing. 12,069,466 (2024)
2. Systems and Methods for Spatial Audio Rendering. US20240107258A1 (2024)
3. Systems and Methods for Loudspeaker Layout Mapping W02023070105A2 (2023)
4. Systems and Methods for Audio Upmixing EP 4252432 (2023)
5. Systems and Methods for Spatial Audio Rendering. 11,722,833 (2023)
6. Systems and Methods for Audio Upmixing WO2022132197A1 (2022)
7. Systems and Methods for Spatial Audio Rendering. 11,206,504 (2021)
8. Systems and Methods for Audio Upmixing US202220400351A1 (2021)
9. Systems and Methods for Spatial Audio Rendering. WO2020206177 A1 (2020)
10. Audio Output Device Specific Audio Processing. 9,860,641 (2018)
11. Audio Content Enhancement Using Bandwidth Extension Techniques. 8,705,764 (2014)
12. Room Acoustic Response Modeling and Equalization with Linear Predictive Coding and Parametric Filters. 8,363,853 (2013)
13. Cross-Over Frequency Selection and Optimization of Response Around Crossover. 8,363,852 (2013)
14. Stereo Expansion with Binaural Modeling. 8,229,143 (2012)
15. Phase Equalization for Multi-Channel Loudspeaker-Room Responses. 8,218,789 (2010)
16. Combined Multirate-Based and FIR-Based Filtering Technique for Room Acoustic Equalization. 8,077,880 (2011)

17. System and Method for Automatic Multiple Listener Room Correction with Low Order Filters. 8,005,228 (2011)
18. Personal Audio Docking Station. D629,391 (2010)
19. Cross-Over Frequency Selection and Optimization of Response Around Cross-Over. 7,826,626 (2010)
20. System and Method for Automatic Room Acoustic Correction in Multi-Channel Audio Environments. 7,769,183 (2010)
21. Phase Equalization for Multi-Channel Loudspeaker-Room Responses. 7,720,237 (2010)
22. System and Method for Automatic Multiple Listener Room Acoustic Correction with Low Filter Orders. 7,567,675 (2009)

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## Selected Press Coverage

- April 16, 2022**     **Joe N Tell:** interview on acoustics and psychoacoustics  
<https://www.youtube.com/watch?v=82QTD0RoB0Q>
- January 10, 2021**     **The New Yorker:** The origins of video-chat voice  
<https://www.newyorker.com/tech/annals-of-technology/the-origins-of-video-chat-voice>
- May 11, 2019**     **Observer:** Inside the world of sound designers who create tech's noisy notifications  
<https://observer.com/2019/05/app-sounds-notifications-design/>
- May 1, 2019**     **New York Magazine:** I tried a \$500 sound-therapy machine  
<https://www.thecut.com/2019/05/i-tried-a-usd500-sound-therapy-machine.html>
- June 28, 2018**     **Sputnik News International:** Will spherical music venues in London and Las Vegas really produce great sound?  
<https://sputniknews.com/20180628/msg-sphere-london-acoustics-1065851435.html>
- December 2, 2017**     **BBC World Service:** Symphony of the Stones  
<http://www.bbc.co.uk/programmes/w3csvq4j>
- February 28, 2017**     **National Herald:** Professors Recreate the Sounds of an Ancient Greek Church in LA  
<https://www.thenationalherald.com/152455/professors-recreate-sounds-ancient-greek-church-la/>
- September 2016**     **USC Trojan Family Magazine:** He uses sound to travel through time  
<https://news.usc.edu/trojan-family/he-uses-sound-to-travel-through-time/>
- March 9, 2016**     **Open Culture:** Museums of Lost Sound  
<http://www.openculture.com/2016/03/mapping-the-sounds-of-greek-byzantine-churches-how-researchers-are-creating-museums-of-lost-sound.html>
- February 26, 2016**     **CBC Radio:** Mapping the Sound of Byzantine Churches  
<http://www.cbc.ca/radio/spark/311-unscrambling-encryption-history-in-3d-and-more-1.3465784/mapping-the-sound-of-byzantine-churches-1.3465806>

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|---------------------------|--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| <b>February 19, 2016</b>  | <b>The Atlantic:</b> Hearing the Lost Sounds of Antiquity<br><a href="http://www.theatlantic.com/technology/archive/2016/02/byzantine-angel-wings/470076/">http://www.theatlantic.com/technology/archive/2016/02/byzantine-angel-wings/470076/</a>                                               |
| <b>September 14, 2015</b> | <b>UCLA Newsroom:</b> Measuring the sound of angels singing<br><a href="http://newsroom.ucla.edu/stories/measuring-the-sound-of-angels-singing">http://newsroom.ucla.edu/stories/measuring-the-sound-of-angels-singing</a>                                                                       |
| <b>February 5, 2015</b>   | <b>Gizmodo:</b> No, women's voices are not easier to understand than men's voices<br><a href="http://gizmodo.com/no-siri-is-not-female-because-womens-voices-are-easier-1683901643">http://gizmodo.com/no-siri-is-not-female-because-womens-voices-are-easier-1683901643</a>                     |
| <b>September 30, 2014</b> | <b>IEEE Spectrum:</b> Why mobile voice quality still stinks and how to fix it<br><a href="http://spectrum.ieee.org/telecom/wireless/why-mobile-voice-quality-still-stinksand-how-to-fix-it">http://spectrum.ieee.org/telecom/wireless/why-mobile-voice-quality-still-stinksand-how-to-fix-it</a> |
| <b>June 21, 2012</b>      | <b>NPR All Things Considered:</b> He'll Retune Your Living Room<br><a href="http://www.npr.org/blogs/therecord/2012/06/21/155501975/hell-retune-your-living-room">http://www.npr.org/blogs/therecord/2012/06/21/155501975/hell-retune-your-living-room</a>                                       |
| <b>Sept. 5, 2011</b>      | <b>New York Times:</b> Sound, the Way the Brain Prefers to hear it<br><a href="http://www.nytimes.com/2011/09/06/science/06sound.html?_r=1&amp;pagewanted=all">http://www.nytimes.com/2011/09/06/science/06sound.html?_r=1&amp;pagewanted=all</a>                                                |
| <b>Nov. 29, 2006</b>      | <b>Widescreen Review:</b> Chris Kyriakakis wins 2006 World Technology Award<br><a href="http://www.widescreenreview.com/news_detail.php?id=12381">http://www.widescreenreview.com/news_detail.php?id=12381</a>                                                                                   |
| <b>Oct. 27, 2006</b>      | <b>USC News:</b> Immersive Sound-The Next Step in Audio<br><a href="http://www.usc.edu/uscnnews/stories/12915.html">http://www.usc.edu/uscnnews/stories/12915.html</a>                                                                                                                           |

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## Journal Publications

1. S. Gerstel, C. Kyriakakis, S. Antonopoulos, K. Raptis, and J. Donahue, "Holy, Holy, Holy: Hearing the Voices of Angels", *Gesta*, 60(1) 2021
2. S. Gerstel, C. Kyriakakis, S. Antonopoulos, K. Raptis, and J. Donahue, "Soundscapes of Byzantium: The Acheiropoietos Basilica and the Cathedral of Hagia Sophia in Thessaloniki," *Hesperia*, Issue 87.1, January-March 2018
3. S. Gerstel, C. Kyriakakis, S. Antonopoulos, K. Raptis, and J. Donahue, "Soundscapes of Byzantium", *Speculum* 92(5), October 2017
4. S. Gerstel and C. Kyriakakis, "Uncovering the Acoustic Mysteries of Byzantine Thessaloniki", *Faith and Form*, 49(3), Oct. 2016
5. C. S. Lin and C. Kyriakakis, "High Definition Audio Spectrum Separation via Frequency Response Masking Filter Banks", *Journal of the Audio Engineering Society*, 57(10), 807-821, (2009)
6. C. S. Lin and C. Kyriakakis, "Removing Additive Noise via Neuro-Fuzzy Based Reinforcement Learning," *Journal of the Acoustical Society of America*, 124(2), 1026-1037, (2008)
7. D. Cantzos and C. Kyriakakis, "Quality Enhancement of Compressed Audio Based on Statistical Conversion," *EURASIP Journal on Audio, Speech, and Music Processing*, vol. 2008, p. 1-15, (May 2008).
8. S. Bharitkar and C. Kyriakakis, "Visualization of Multiple Listener Room Acoustic Equalization with Sammon Map," *IEEE Transactions on Speech and Audio Processing*, 15(2), 542-551, (2007).
9. P. G. Georgiou and C. Kyriakakis, "Robust Maximum Likelihood Source Localization: The Case for Sub-Gaussian versus Gaussian," *IEEE Transactions on Speech and Audio Processing*, 14(4), 1470-1480, (2006).
10. P. G. Georgiou and C. Kyriakakis, "Maximum Likelihood Parameter Estimation under Impulsive Conditions, a Sub-Gaussian Signal Approach," *EURASIP Signal Processing Journal*, 10, 3061-3075 (2006).

11. A. Mouchtaris, S. S. Narayanan, and C. Kyriakakis, "Multichannel Audio Synthesis by Subband-Based Spectral Conversion and Parameter Adaptation," *IEEE Trans. Speech and Audio Processing*, 13(2), 263-274, (2005).
12. S. Bharitkar and C. Kyriakakis, "Robustness of Spatial Average Equalization: A Statistical Reverberation Model Approach," *Journal of the Acoustical Society of America*, 116(6), 3491-3497, (2004).
11. R. Zimmermann, C. Kyriakakis, C. Shahabi, C. Papadopoulos, A. A. Sawchuk, U. Neumann, "The Remote Media Immersion System: Internet Meets the Future Home Theater," *IEEE Multimedia Magazine*, March 2004.
12. S. Bharitkar and C. Kyriakakis, "Selective Signal Cancellation For Multiple-Listener Audio Applications Using Eigenfilters," *IEEE Transactions on Multimedia*, vol. 5(3), 329-338, (2003).
13. D. Yang, H. Ai, C. Kyriakakis, and C. -C. Jay Kuo, "High-Fidelity Multichannel Audio Coding With Karhunen-Loeve Transform," *IEEE Transactions on Speech and Audio Processing*, 11(4), 365-380, (2003).
14. A. Mouchtaris, S. S. Narayanan, and C. Kyriakakis, "Virtual Microphones for Multichannel Audio Resynthesis," *EURASIP Journal on Applied Signal Processing*, Special Issue on Digital Audio for Multimedia Communications, 10, 968-979, (2003).
15. D. Yang. Ai, C. Kyriakakis and C.-C. Jay Kuo, "Progressive Syntax-Rich Audio Coding for Multichannel Audio Sources," *EURASIP Journal on Applied Signal Processing*, Special Issue on Digital Audio for Multimedia Communications, 10, 980-992, (2003).
16. A. Mouchtaris, P. Reveliotis, and C. Kyriakakis, "Inverse Filter Design for Immersive Audio Rendering over Loudspeakers," *IEEE Transactions on Multimedia*, 2(2), 77-87, (2000).
17. P. Georgiou, C. Kyriakakis, and P. Tsakalides, "Alpha Stable Modeling of Noise and Robust Time-Delay Estimation in the Presence of Impulsive Noise," *IEEE Transactions on Multimedia*, 1(3), 291-301, (1999).
18. C. Kyriakakis, P. Tsakalides, and T. Holman, "Surrounded by Sound: Immersive Audio Acquisition and Rendering Methods," *IEEE Signal Processing Magazine*, 16(1), 55-66, (1999).

19. C. Kyriakakis, T. Holman, J.-S. Lim, H. Hong, and H. Neven, "Signal Processing, Acoustics, and Psychoacoustics for High Quality Desktop Audio," *Journal of Visual Communication and Image Representation*, **9**(1), 51-61, (1998).
20. C. Kyriakakis, "Fundamental and Technological Limitations of Immersive Audio Systems," *Proceedings of the IEEE*, **86**(5), 941-951, (1998).

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### Technical Conference Publications

1. Wang, Z., Kensek, K., Kyriakakis, C., Zyda, M., and Narhi, E. "Augmented Reality in Room Acoustics: A simulation tool for mobile devices with auditory feedback", 7th International Conference on Architecture and Built Environment (April 2020)
2. D. Cantzos, A. Mouchtaris, and C. Kyriakakis, "Perceptually-Driven Scalable MDCT Enhancement of Compressed Audio Based on Statistical Conversion", *Proceedings of IEEE International Symposium on Multimedia*, Dec. 2011.
3. S. Bharitkar and C. Kyriakakis, "Comparison between Spherical Headmodels and HRTFs in Upmixing for Headphone Based Virtual Surround and Stereo Expansion", *Proceedings of the AES 129th Convention in San Francisco, CA*, November 4-7, 2010.
4. C. S. Lin, "Design of Phase Equalizer for Rational Filter Banks Using Minimax Approach", *IEEE International Symposium on Industrial Electronics*, June 2009.
5. D. Cantzos and C. Kyriakakis, "Bandwidth Extension of Low Bitrate Compressed Audio Based on Statistical Conversion", *Proceedings of IEEE ICME 2009*.
6. S. Bharitkar, C. Kyriakakis, T. Holman, "Variable-Octave Complex Smoothing for Loudspeaker-Room Response Equalization", *IEEE International Conference on Consumer Electronics (ICCE 2008)*, Las Vegas, NV, 9-13 Jan., 2008.
7. D. Cantzos, A. Mouchtaris, and C. Kyriakakis, "Synthesis of Enhanced Audio from Low Bitrate Compressed Audio Based on Unit Selection and Statistical Conversion

- Methods,” 42<sup>nd</sup> Asilomar Conference on Signals, Systems, and Computers, Pacific Grove, CA, Oct. 26, 2008.
8. S. Bharitkar, Y. Zhang, and C. Kyriakakis, “Room Acoustic Response Modeling and Equalization with Linear Predictive Coding and Parametric Filters for Speech and Audio Enhancement,” in *Proc. 40th IEEE Asilomar Conf. on Signals, Systems, & Computers*, Pacific Grove (CA), Oct. 2006.
  9. S. Bharitkar and C. Kyriakakis, “Cascaded FIR Filters for Multiple Listener Low Frequency Room Acoustic Equalization,” accepted in *2006 IEEE Intl. Conf. Acoust., Speech, & Signal Proc. (ICASSP-2006)*, Toulouse, France, May 2006.
  10. C. -S. Lin and C. Kyriakakis, “A Morphing Approach for Synthesizing Multichannel Recordings,” 39th IEEE Asilomar Conference on Signals, Systems, & Computers, Pacific Grove, CA, November, 2005.
  11. J. Peterson and C. Kyriakakis, “Analysis of Fast Localization Algorithms for Acoustical Environments,” 39th IEEE Asilomar Conference on Signals, Systems, & Computers, Pacific Grove, CA, November, 2005.
  12. S. Bharitkar and C. Kyriakakis, “Comparison Between Time Delay Based and Nonuniform Phase Based Equalization for Multichannel Loudspeaker-Room Responses,” 39th IEEE Asilomar Conference on Signals, Systems, & Computers, Pacific Grove, CA, November, 2005.
  13. S. Bharitkar and C. Kyriakakis, “Optimization of the Bass Management Filter Parameters for Multichannel Audio Applications,” 39th IEEE Asilomar Conference on Signals, Systems, & Computers, Pacific Grove, CA, November, 2005.
  14. S. Bharitkar and C. Kyriakakis, “Objective Function for Automatic Multiposition Equalization and Bass Management Filter Selection,” Audio Engineering Society, New York, October, 2005.
  15. S. Sundaram and C. Kyriakakis, “Phantom Audio Sources With Vertically Separated Speakers,” Audio Engineering Society, New York, October, 2005.
  16. S. Bharitkar and C. Kyriakakis, “Optimization of Crossover Frequency and Crossover Region Response for Multichannel Acoustic Applications,” 13th European

- Signal Processing Conference (EUSIPCO 2005), Antalya, Turkey, September, 2005.
17. J. Peterson and C. Kyriakakis, "Hybrid Algorithm for Robust, Real-Time Source Localization in Reverberant Environments," International Conference on Acoustics, Speech, and Signal Processing (ICASSP '05), Philadelphia, March 2005.
  18. J. Peterson and C. Kyriakakis, "Choosing Candidate Locations for Source Localization," Joint Workshop on Hands-Free Speech Communication and Microphone Arrays (HSCMA '05), Rutgers, March 2005.
  19. C. Papadopoulos, C. Kyriakakis, A. A. Sawchuk, X. He, "Cyberseer: 3D Audio-Visual Immersion for Network Security and Management," Workshop on Visualization and Data Mining for Computer Security (VizSEC/DMSEC) 2004, Fairfax, VA, October, 2004.
  20. H. Huang and C. Kyriakakis, "Binaural Noise Reduction Combining Binaural Analysis and Spectral Subtraction," Proc. 38th IEEE Asilomar Conference on Signals, Systems, & Computers, Pacific Grove, CA. Nov. 2004.
  21. H. Huang and C. Kyriakakis, "Computationally Efficient Blind Dereverberation of Audio Signals," Proceedings of the 117<sup>th</sup> Convention of the Audio Engineering Society, New York, October, 2004.
  22. M. Boueri and C. Kyriakakis, "Audio Signal Decorrelation based on a Critical Band Approach," Proceedings of the 117<sup>th</sup> Convention of the Audio Engineering Society, New York, October, 2004.
  23. S. Bharitkar and C. Kyriakakis, "Nonlinear Projection Algorithm for Evaluating Multiple Listener Equalization Algorithm Performance," Proceedings of the 117<sup>th</sup> Convention of the Audio Engineering Society, New York, October, 2004.
  24. R. Sadek and C. Kyriakakis, "A Novel Multichannel Panning Algorithm for Standard and Arbitrary Loudspeaker Configurations," Proceedings of the 117<sup>th</sup> Convention of the Audio Engineering Society, New York, October, 2004.
  25. A.A. Sawchuk, E. Chew, R. Zimmermann, C. Papadopoulos and C. Kyriakakis, "Distributed Immersive Performance," Corporation for Education Network Initiatives in California (CENIC) Eighth Annual Conference, March 15-17, 2004, Marina del Rey, CA.

26. C. S. Lin and C. Kyriakakis, "Multi-Frequency Noise Removal Based on Reinforcement Learning," Preprint No. 5966, presented at the 115th AES Convention, New York, NY, October, 2003.
27. S. Bharitkar, P. Hilmes, C. Kyriakakis, "Sensitivity of Multichannel Room Equalization to Listener Position," in Proceedings of the AES 115<sup>th</sup> Convention, NY, Oct 2003.
28. H. Huang and C. Kyriakakis, "Subband Adaptive Filtering For Acoustic Noise Reduction," in Proceedings of the AES 115th Convention, NY, Oct 2003.
29. S. Bharitkar and C. Kyriakakis, "The Influence of Reverberation on Multichannel Equalization: An Experimental Comparison Between Methods," Proc. 37th IEEE Asilomar Conference on Signals, Systems, & Computers, Pacific Grove, CA. Nov. 2003. **(Best Paper Award)**
30. J. M. Peterson, P. Georgiou, and C. Kyriakakis, "The Use of Fractional Lower-Order Statistics in acoustical environments," in Proceedings of Asilomar Conference on Signals, Systems, and Computers, 2003.
31. C. S. Lin and C. Kyriakakis, "Frequency Response Masking Approach for Designing Filter Banks with Rational Sampling Factors," IEEE WASPAA 2003, Mohonk, NY, October, 2003.
32. S. Bharitkar and C. Kyriakakis, "A Comparison Between Multi-channel Audio Equalization Filters Using Warping," 2003 IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA 2003), New Paltz, NY. Oct. 2003.
33. S. Bharitkar and C. Kyriakakis, "Sensitivity of multichannel room equalization to listener position," Proc. IEEE Conf. on Multimedia & Expo (ICME), Baltimore. July 2003.
34. S. Bharitkar, P. Hilmes, C. Kyriakakis, "Analysis of Multiple Listener Equalization Performance Due to Displacement Effects," 145th Meeting of the Acoustical Soc. America. Nashville (TN). May 2003.
35. A.A. Sawchuk, C. Kyriakakis, R. Zimmermann, C. Papadopoulos, C. Shahabi, U. Neumann and T. Holman, "Remote Media Immersion," Corporation for Education Network Initiatives in California (CENIC) Seventh Annual Conference, May 7-9, 2003, Santa Barbara, CA.

36. S. Rishi, C. Papadopoulos, and C. Kyriakakis, "Loss Concealment for Multichannel Streaming Audio," Network and Operating System Support for Digital Audio and Video (NOSSDAV '03), Monterey, CA, June, 2003.
37. P. G. Georgiou and C. Kyriakakis, "A new model of source dependent noise for robust array signal processing," in Second IEEE Sensor Array and Multichannel Signal Processing Workshop, Aug. 2002.
38. P. G. Georgiou and C. Kyriakakis, "Signal Parameter Estimation and Localization via Maximum Likelihood Using and Sensor Array in the Presence of Levy Noise," ICSP 2002, Beijing.
39. S. Bharitkar and C. Kyriakakis, "Perceptual Multiple Location Equalization with Clustering," in Proc. 36th IEEE Asilomar Conference on Signals, Systems, & Computers, Pacific Grove, CA, Nov. 2002.
40. S. Bharitkar, P. Hilmes, and C. Kyriakakis, "Robustness of Spatial Averaging Equalization Methods: A Statistical Approach," in Proc. 36th IEEE Asilomar Conference on Signals, Systems, & Computers, Pacific Grove, CA, Nov. 2002.
41. G. Georgiou and C. Kyriakakis, "A novel model for reverberant signals; robust maximum likelihood localization of real signals based on a Sub-Gaussian model," in Proc. 36th IEEE Asilomar Conference on Signals, Systems, & Computers, Pacific Grove, CA, Nov. 2002.
42. A. Mouchtaris, S. S. Narayanan, and C. Kyriakakis, "Maximum Likelihood Constrained Adaptation for Multichannel Audio Synthesis," in Proc. 36th IEEE Asilomar Conference on Signals, Systems, & Computers, Pacific Grove, CA, Nov. 2002.
43. S. Bharitkar and C. Kyriakakis, "Visualization of Multiple Listener Room Response Equalization using Sammon Map," in Proc. IEEE Intl. Conf. on Multimedia & Expo (ICME 2002), Lausanne, Switzerland, Aug. 2002.
44. A. Mouchtaris, S. S. Narayanan, and C. Kyriakakis, "Multiresolution Spectral Conversion for Multichannel Audio Resynthesis," in Proc. IEEE International Conference on Multimedia and Expo (ICME 2002), vol. 2, p. 273-276, Lausanne, Switzerland, Aug. 2002.
45. A. Mouchtaris, S. S. Narayanan, and C. Kyriakakis, "Efficient Multichannel Audio Resynthesis by Subband-

- Based Spectral Conversion,” in Proc. European Processing Conference (EUSIPCO), vol. 1, p. 413-416, Toulouse, France, Sept. 2002.
46. P. G. Georgiou and C. Kyriakakis, “A robust array signal processing maximum likelihood estimator based on sub-gaussian signals,” in Proc. European Processing Conference (EUSIPCO), Toulouse, France, Sept. 2002.
  47. D. Yang, H. Ai, C. Kyriakakis, and C.-C. Jay Kuo, “Design of Progressive Syntax-Rich Multichannel Audio Codec,” Proc. SPIE Vol. 4674, p. 121-123, Media Processors 2002, San Jose, CA, January, 2002.
  48. D. Yang, H. Ai, C. Kyriakakis and C.-C. Jay Kuo, “Error-Resilient Design of High Fidelity Scalable Audio Coding,” Proc. SPIE Vol. 4740, Digital Wireless Communications IV, Orlando, Florida USA, April 1-5, 2002.
  49. J. M. Peterson and C. Kyriakakis, “Using subband filters to reduce the complexity of real-time signal processing,” Preprint No. 5663, presented at the 113th AES Convention, Los Angeles, CA, October, 2002.
  50. D. Yang, H. Ai, C. Kyriakakis, and C.-C. Jay Kuo, “Design of Error-Resilient Layered Audio Codec,” Preprint No. 5681, presented at the 113th AES Convention, Los Angeles, CA, October, 2002.
  51. S. Bharitkar, P. Hilmes, and C. Kyriakakis, “A Statistical Analysis of the Robustness of Room Equalization Using Spatial Averaging,” Preprint No. 5669, presented at the 113th AES Convention, Los Angeles, CA, October, 2002.
  52. P. G. Georgiou and C. Kyriakakis, “A Novel Method for Source Localization using Microphone Arrays in Reverberant Environments,” Preprint No. 5643, presented at the 113th AES Convention, Los Angeles, CA, October, 2002.
  53. A. Mouchtaris, S. S. Narayanan, and C. Kyriakakis, “Gaussian-Mixture-Model-Based Methods for Multichannel Audio Resynthesis,” Preprint No. 5647, presented at the 113th AES Convention, Los Angeles, CA, October, 2002.
  54. C.-S. Lin and C. Kyriakakis, A Fuzzy Cerebellar Model Approach for Synthesizing Multichannel Recordings,” Preprint No. 5675, presented at the 113th AES Convention, Los Angeles, CA, October, 2002.
  55. S. Bharitkar and C. Kyriakakis, “A Cluster Centroid Method for Room Response Equalization at Multiple

- Locations,” IEEE WASPAA 2001, Mohonk, NY (WASPAA: Workshop on Applications of Signal Processing to Audio and Acoustics).
56. J. –S. Lim and C. Kyriakakis, “Multirate Adaptive Filtering for Immersive Audio,” ICASSP ’01, Salt Lake City, Utah, May, 2001 (ICASSP: International Conference on Acoustics, Speech, and Signal Processing).
  57. D. Yang, H. Ai, C. Kyriakakis, and C. –C. Kuo, “Adaptive Karhunen-Loeve Transform for Enhanced Multichannel Audio Coding, SPIE, San Diego, 2001.
  58. Yang, H. Ai, C. Kyriakakis, and C. –C. Kuo, “Embedded High-Quality Multichannel Audio Coding,” Conference on Media Processors, Symposium on Electronic Imaging 2001, San Jose, CA, January 21-26, 2001.
  59. S. Bharitkar and C. Kyriakakis, “A Classification Scheme for Room Acoustical Responses,” ICASSP 2001, Salt Lake City, Utah.
  60. J. –S. Lim and C. Kyriakakis, “Adaptive Synthesis of Immersive Audio Rendering Filters,” 111<sup>th</sup> AES Convention, New York, NY, 2001.
  61. S. Bharitkar and C. Kyriakakis, “New Factors in Room Equalization Using a Fuzzy Logic Approach, , 111<sup>th</sup> AES Convention, New York, NY, 2001.
  62. A. Mouchtaris and C. Kyriakakis, “Time Frequency Methods for Virtual Microphone Signal Synthesis,” 111<sup>th</sup> AES Convention, New York, NY, 2001.
  63. S. Bharitkar and C. Kyriakakis, “Eigenfilters for Signal Cancellation,” ISPACS 2000, Hawaii.
  64. C. Kyriakakis and A. Mouchtaris, “Virtual Microphones for Multichannel Audio Applications,” ICME 2000, New York, July 2000.
  65. S. Bharitkar and C. Kyriakakis, “Selective Signal Cancellation for Multiple Listener Audio Applications: An Information Theory Approach,” ICME 2000, New York, July 2000.
  66. P. Georgiou and C. Kyriakakis, “A Multiple Input Single Output Model for Immersive Audio Rendering in Real Time,” ICME 2000, New York, July 2000.
  67. S. Rizzo, J. G. Buckwalter, L. Humphrey, C. van der Zaag, T. Bowerly, C. Chua, U. Neumann, and C. Kyriakakis, “A Virtual Reality Classroom Scenario for ADHD Assessment,” The 28th Annual Conference of the

- International Neuropsychological Society, Denver, Colorado, February, 2000.
68. P. G. Georgiou and C. Kyriakakis, "Modeling of Head-Related Transfer Functions for Immersive Audio Using a State-Space Approach," 33<sup>rd</sup> Asilomar Conference on Signals, Systems, and Computers, Pacific Grove, California, October, 1999.
  69. A. Mouchtaris, Z. Zhu, and C. Kyriakakis, "High-Quality Internet Audio over ATM Networks," 33<sup>rd</sup> Asilomar Conference on Signals, Systems, and Computers, Pacific Grove, California, October, 1999.
  70. C. Kyriakakis, S. Zeadally, and T. Holman, "Multichannel Audio over Internet2 Using ATM," Audio Engineering Society, preprint no. 5000, (1999).
  71. C. Kyriakakis, "Immersive Audio Rendering on the TI C62 DSP Platform," TI DSPFest, Houston, TX, August, 1999
  72. P. G. Georgiou, P. Tsakalides, and C. Kyriakakis, "Alpha-Stable Robust Modeling of Background Noise for Enhanced Sound Source Localization," ICASSP '99, Phoenix, Arizona, March, 1999.
  73. A. Mouchtaris, P. Reveliotis, and C. Kyriakakis, "Non-minimum Phase Inverse Filtering Methods for Immersive Audio Rendering," ICASSP '99, Phoenix, Arizona, March, 1999.
  74. C. Kyriakakis and T. Holman, "Video-Based Head Tracking for Improvements in Multichannel Loudspeaker Audio," Audio Engineering Society, preprint no. 4845, (1998).
  75. A. Mouchtaris, J.-S. Lim, T. Holman, and C. Kyriakakis, "Head-Related Transfer Function Synthesis for Immersive Audio," Proceedings of the IEEE Multimedia Signal Processing Workshop (MMSP '98), Redondo Beach, California, December, 1998.
  76. C. Kyriakakis and T. Holman, "Immersive Audio for Desktop Systems," Acoustical Society of America International Conference on Acoustics, Seattle Washington, June, 1998
  77. C. Kyriakakis and T. Holman, "Immersive Audio for the Desktop: Signal Processing Requirements," ICASSP '98, Seattle, Washington, May, 1998.

78. A. Mouchtaris, J.-S. Lim, T. Holman, and C. Kyriakakis, "Signal Processing Considerations for Immersive Audio Rendering," Proceedings of the 10<sup>th</sup> Tyrrhenian Conference on Multimedia Communications, Ischia, Italy, (1998).
79. P. Georgiou, C. Kyriakakis, and P. Tsakalides "Microphone Arrays for Speaker Direction Identification," Proceedings of the IEEE on Applications of Signal Processing to Audio and Acoustics, WASPAA, (1997).
80. T. Holman and C. Kyriakakis, "Acoustics and Psychoacoustics of Desktop Systems," 134th meeting of the Acoustical Society of America, San Diego, CA, (1997).