

UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

ROKU, INC.
Petitioner

v.

DOLBY INTERNATIONAL AB,
Patent Owner

Case IPR2024-01354
U.S. Patent No. 6,339,757

**DECLARATION OF DR. TIM A. WILLIAMS IN SUPPORT OF
PETITION FOR *INTER PARTES* REVIEW OF U.S. PATENT 6,339,757**

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Patent Trial and Appeal Board
U.S. Patent & Trademark Office
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Roku, Inc. EX1003
U.S. Patent No. 6,339,757

TABLE OF CONTENTS

I.	Introduction.....	1
II.	Qualifications.....	2
III.	Materials considered	5
IV.	Relevant legal standards.....	7
	A. My understanding of level of ordinary skill	7
	B. My understanding of claim construction	8
	C. My understanding of obviousness	9
V.	Background of the Technology	10
	A. General background of digital audio signal encoding.....	11
	B. Quantization of signals for bit allocation was well known prior to the '757 patent.	16
	C. Dividing a frequency spectrum into frequency bands and assigning bits to those bands was well known prior to the '757 patent.	19
	D. Assigning bits based on psychoacoustic factors was a well- known technique prior to the '757 patent.	23
	E. Assigning bits using a two-step bit allocation process was a well-known technique prior to the '757 patent.	25
VI.	Overview of the '757 patent	27
	A. The '757 patent's bit allocation scheme	28
	B. The patent claims	36
	C. The prosecution history of the '757 patent	37
VII.	CLAIM CONSTRUCTION	41
VIII.	GROUND 1: CLAIMS 1-3, 6-7, 10-11, 14, AND 16 of the '757 Patent are Obvious Over The Combination of Mahieux and Fielder.....	44
	A. Overview of Mahieux	44
	B. Overview of Fielder	49
	C. A POSA would have been motivated to improve Mahieux's bit allocation method using Fielder's second bit allocation step.	52
	D. Mahieux in view of Fielder renders claim 1 obvious.....	58

Declaration of Dr. Tim A. Williams
U.S. Patent No. 6,339,757

1.	Mahieux discloses “[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:” [1P].	58
2.	Mahieux discloses “a step for dividing at least one frequency interval into a plurality of frequency-units;” [1A].	60
3.	Mahieux discloses, or at least renders obvious, “a step for obtaining a representative of each frequency-unit obtained;” [1B].	62
4.	Mahieux discloses, or at least renders obvious, “a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria;” [1C].	64
5.	Mahieux discloses, or at least 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; and” [1D].	69
6.	Mahieux in view of Fielder teaches “a second bit allocation step for determining further bit allocation for each of said available frequency-units based on psychoacoustic criteria.” [1E].	73
E.	Mahieux in view of Fielder renders claim 2 obvious.	77
F.	Mahieux in view of Fielder renders claims 3, 7, 11, and 14 obvious.	80
G.	Mahieux in view of Fielder renders claim 6 obvious.	82
H.	Mahieux in view of Fielder renders claim 10 obvious.	84
I.	Mahieux in view of Fielder renders claim 16 obvious.	86
IX.	GROUND 2: Claims 1-17 of the ’757 Patent is Obvious Over The Combination of Mahieux, Fielder, and Akagiri.	87
A.	Overview of Akagiri	87
B.	Motivation to combine Akagiri with Mahieux and Fielder	88
C.	Mahieux in view of Fielder and Akagiri renders claims 4, 8, 12, and 15 obvious.	92

Declaration of Dr. Tim A. Williams
U.S. Patent No. 6,339,757

D.	Mahieux in view of Fielder and Akagiri renders claims 5, 9, 13, and 17 obvious.....	94
E.	Mahieux in view of Fielder and Akagiri renders claims 1-3, 6-7, 10-11, 14, and 16 obvious.....	95
X.	GROUND 3: Claims 1-3, 6-7, 10-11, 14, AND 16 of the '757 Patent are Obvious Over The Combination of Veldhuis and Galand.....	96
A.	Overview of Veldhuis	96
B.	Overview of Galand.....	101
C.	A POSA would have been motivated to use Galand's mathematical model in determining Veldhuis's initial bit allocation.	102
D.	Veldhuis in view of Galand renders claim 1 obvious.	105
1.	Veldhuis discloses, or at least renders obvious, "[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:" [1P].	105
2.	Veldhuis discloses "a step for dividing at least one frequency interval into a plurality of frequency-units;" [1A].	106
3.	Veldhuis discloses "a step for obtaining a representative of each frequency-unit obtained;" [1B].	108
4.	Veldhuis, alone or in combination with Galand, renders obvious "a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria;" [1C].	109
5.	Veldhuis in view of Galand 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; and;" [1D].	114
6.	Veldhuis discloses, or at least renders obvious, "a second bit allocation step for determining further bit allocation for each of said available frequency-units based on psychoacoustic criteria." [1E].	117

Declaration of Dr. Tim A. Williams
U.S. Patent No. 6,339,757

E.	Veldhuis in view of Galand renders claim 2 obvious.	122
F.	Veldhuis in view of Galand renders claims 3, 7, 11, and 14 obvious.	123
G.	Veldhuis in view of Galand renders claim 6 obvious.	124
H.	Veldhuis in view of Galand renders claim 10 obvious.	126
I.	Veldhuis in view of Galand renders claim 16 obvious.	127
XI.	GROUND 4: Claims 1-17 of the '757 Patent are Obvious Over The Combination of Veldhuis, Galand, and Reaves.	128
A.	Overview of Reaves.....	128
B.	A POSA would have combined Reaves with Veldhuis and Galand.	129
C.	Veldhuis in view of Galand and Reaves renders claims 4, 8, 12, and 15 obvious.....	132
D.	Veldhuis in view of Galand and Reaves renders claims 5, 9, 13, and 17 obvious.....	133
E.	Veldhuis in view of Galand and Reaves renders claims 1-3, 6-7, 10-11, 14, and 16 obvious.....	134
XII.	Conclusion.....	136

I, Tim A. Williams, declare as follows:

I. INTRODUCTION

1. I have been retained on behalf of Roku, Inc. in the above-captioned *inter partes* review proceeding to provide my expert opinions and knowledge. I understand that this proceeding involves U.S. Patent No. 6,339,757 ('757 patent), titled "Bit Allocation Method for Digital Audio Signals," by Do Hui Teh et al., and that the '757 patent is currently assigned to Dolby International AB (Patent Owner).

2. In preparing this declaration, I have reviewed and am familiar with the '757 patent and its prosecution history, as well as all references cited herein. I confirm that, to the best of my knowledge, the accompanying exhibits are true and accurate copies of what they purport to be, and that an expert in the field would reasonably rely on them to formulate opinions such as those set forth in this declaration.

3. I am familiar with the technology described in the '757 patent and the state of the art as of the patent's earliest possible priority date, February 19, 1993. I have been asked to provide my independent technical review, analysis, insights, and opinions regarding the '757 patent and the references that form the basis for the grounds of unpatentability set forth in the petition.

4. I am being compensated at my rate of \$750 per hour for my work on

this case. My compensation is not dependent upon my opinions, testimony, or the outcome of this case.

II. QUALIFICATIONS

5. The details of my education, work experience, research, and publications are summarized in my curriculum vitae, attached hereto as Exhibit 1004. I provide here a summary of some of the relevant information from my curriculum vitae, as well as other relevant background information.

6. I earned a Bachelor of Science degree in Electrical Engineering (BSEE) from Michigan Technological University in 1976. In 1982, I earned a Master's degree, also in Electrical Engineering (MSEE), from the University of Texas at Austin. I earned a Doctor of Philosophy (Ph.D.), also from the University of Texas at Austin, in 1985. My doctoral dissertation was titled "Digital Signal Processing Techniques for Acoustic Log Data." In 1991, I earned a Master of Business Administration degree from the University of Texas at Austin.

7. I have more than 40 years of professional experience in speech compression, wireless communications, and telecom technology, including in large corporations and start-ups, as well as through consulting work.

8. For example, from 1979 through 1991, I held the positions of Senior Engineer, Senior Member of the Technical Staff, and Member of the Technical Staff at Motorola, Inc. My work at Motorola, Inc. included serving as business

Declaration of Dr. Tim A. Williams
U.S. Patent No. 6,339,757

manager, project leader, and senior technical member for projects relating to the development of various cellular modems, transceivers, and transcoders.

9. During that time period, I designed an integrated circuit for speech compression and expansion using Continuously Variable Slope Delta (CVSD) modulation, which was incorporated into Motorola's two-way radios for land mobile radio systems.

10. Additionally, I helped develop speech compression technology at Motorola. My work involved designing integrated circuits and software for Motorola's speech chip, which used LPC and was comparable to the chip used in Texas Instruments' Speak & Spell, famously featured in the movie *E.T.* Additionally, I designed an integrated circuit for Motorola that provided Adaptive Differential Pulse Code Modulation (ADPCM) codec functionality for telecom frame processing.

11. Throughout the 1980s and 1990s, I participated in various standards organizations dedicated to creating speech coding standards. This included involvement with the Telecommunications Industry Association's (TIA) standard for the ADPCM codec. I also served on committees developing speech codecs for second- and third-generation cellular radio standards. My responsibilities extended to designing integrated circuits and software for codecs used in these cellular standards, which contributed to the advancement of mobile communication

technology.

12. In 1991, I co-founded Wireless Access, Inc., a start-up company focused on the Narrow Band PCS equipment market and which developed over-the-air protocols, subscriber equipment, and integrated circuits to deploy two-way paging services. I held the positions of Co-Founder, Chief Technical Officer, Vice President of Engineering, and Vice President of Business Strategy at Wireless Access, Inc. until the company was sold to Glenarye Electronics in 1998.

13. Between 1998 and 2000, I was the Chief Technology Officer and Advisory Board Member of Picazo Communications, which built and sold software PBXs Telephony equipment using VoIP and Circuit Switched Technologies.

14. In 2004, I founded DoceoTech Inc., which provides training for engineers in Wireless, Networking, and Telephony technologies. Since 2004, I have served as the Chairman of DoceoTech.

15. My work in speech coding has continued in recent years. For example, I have provided expert testimony in patent cases involving fourth- and fifth-generation cellular standards, which incorporate advanced speech codecs.

16. I hold more than 25 U.S. patents in wireless and signal-processing technology, as well as numerous patent applications in the same field.

Additionally, I am a Patent Agent registered with the U.S. Patent and Trademark

Office since January 2002.

17. Based on my work and experience, I am qualified to comment on the knowledge of a person of ordinary skill in the art prior to the '757 patent and prior to February 19, 1993.

III. MATERIALS CONSIDERED

18. In formulating my opinions, I have relied upon my training, knowledge, and experience that are relevant to the '757 patent. I have also considered the documents listed below in addition to any other documents cited in this declaration:

Exhibit #	Description
1001	U.S. Patent No. 6,339,757 ("’757 patent")
1002	Prosecution History of U.S. Patent No. 6,339,757 ("’757 Prosecution History")
1005	U.S. Patent No. 5,185,800 to Mahieux ("Mahieux")
1006	U.S. Patent No. 5,109,417 to Fielder et al. ("Fielder")
1007	U.S. Patent No. 5,241,603 to Akagiri et al. ("Akagiri")
1008	U.S. Patent No. 4,896,362 to Veldhuis et al. ("Veldhuis")
1009	U.S. Patent No. 4,677,671 to Galand et al. ("Galand")
1010	U.S. Patent No. 5,579,431 to Reaves ("Reaves")
1011	U.S. Patent No. 5,105,463 to Veldhuis et al. ("Veldhuis’463")
1012	European Patent Publication No. 0145788A1 to Akagiri ("Akagiri’788")
1013	European Patent Publication No. 0525809A2 to Akagiri ("Akagiri’809")
1014	U.S. Patent No. 4680797 to Benke ("Benke")
1015	U.S. Patent No. 4,949,383 to Koh et al. ("Koh")
1016	International Publication No. WO 92/12607 to Davis et al. ("Davis")

Declaration of Dr. Tim A. Williams
U.S. Patent No. 6,339,757

Exhibit #	Description
1017	U.S. Patent No. 4,538,234 to Honda et al. (“Honda”)
1018	U.S. Patent No. 4,790,015 to Callens et al. (“Callens”)
1019	U.S. Patent No. 4,964,166 to Wilson (“Wilson”)
1020	U.S. Patent No. 5,235,671 to Mazor (“Mazor”)
1021	U.S. Patent No. 5,263,088 to Hazu (“Hazu”)
1022	International Publication No. WO 89/003140 to Vössing et al. (“Vössing”)
1023	International Publication No. WO 89/09965 to McLaughlin et al. (“Mclaughlin”)
1024	A 128 kb/s Hi-Fi Audio CODEC Based on Adaptive Transform Coding with Adaptive Block Size MDCT (“Iwadare”)
1025	Transform coding of audio signals using correlation between successive transform blocks (“Mahieux NPL”)
1026	Predictive SVD-transform coding of speech with adaptive vector quantization (“Masgrau”)
1027	An optimal bit allocation algorithm for sub-band coding (“Westerink”)
1028	Efficient bit allocation for an arbitrary set of quantizers (“Shoham”)
1029	Quantization and bit allocation in speech processing (“Gray”)
1030	U.S. Patent No. 5,151,941 to Nishiguchi et al. (“Nishiguchi”)
1031	Low bit-rate coding of high-quality audio signals: an introduction to the MASCAM system (“Theile”)
1032	Subband coding of high-fidelity quality audio signals at 128 kbps (“Teh”)
1033	Excerpts from Jeffrey Hass, “Introduction to Computer Music. Chapter Five: Digital Audio. 6. Quantizing, Approximation Errors and Sample Size,” https://cmtext.indiana.edu/index.php , Chapter 5, Section 6 accessed at https://cmtext.indiana.edu/digital_audio/chapter5_quantize.php
1034	Complaint in Dolby Licensing Corv. Roku, Inc., No. 5:24-cv-04660 (N.D. Cal.)

Declaration of Dr. Tim A. Williams
U.S. Patent No. 6,339,757

Exhibit #	Description
1035	Claim Chart for U.S. Pat. No. 6,339,757 and Dolby Digital (ATSC Doc. A/52:2018)
1036	U.S. Patent No. 4,216,354 to Esteban et al. (“Esteban”)
1037	ATSC Standard: Digital Audio Compression (AC-3, E-AC-3), Doc. A/52:2018 (Jan. 25, 2018)
1038	U.S. Patent No. 4,672,457 to Hyatt (“Hyatt”)
1040	Excerpts from Jayant, N. S., Noll, P. “Digital Coding of Waveforms,” Chapters 1, 2, 4, 11, and 12, Prentice-Hall, Inc. (1984)
1041	Certified Translation of International Publication No. WO 89/003140 to Vössing et al. (“Vössing Translation”)
1042	U.S. Patent No. 5,381,145 to Allen et al. (“Allen”)

IV. RELEVANT LEGAL STANDARDS

A. My understanding of level of ordinary skill

19. I understand that a person of ordinary skill in the relevant art is presumed to be aware of all pertinent art, thinks along conventional wisdom in the art, and is a person of ordinary creativity.

20. I have been asked to consider the level of ordinary skill in the field that someone would have had at the time the claimed invention was made. In deciding the level of ordinary skill, I considered the following:

- the levels of education and experience of persons working in the field;
- the types of problems encountered in the field; and
- the sophistication of the technology.

21. My opinion below explains how a person of ordinary skill in the art would have understood the technology described in the references I have identified around the February 1993 timeframe.

22. Based on the disclosure of the '757 patent, a person of ordinary skill in the art (POSA) would have had a Bachelor of Science degree (BS) 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.

23. By related fields, I mean that the required levels of educational and industry experience are on a sliding scale relative to one another. For example, a POSA could have a more advanced educational degree and less industry experience.

24. Regardless of whether I use “I” or a “POSA” in my technical analysis below, all of my statements and opinions are based on how a POSA would have understood or read a document.

B. My understanding of claim construction

25. I have been informed by Petitioner’s counsel that before the Patent Trial and Appeal Board, claims are to be given their ordinary and customary meaning in light of the specification as would have been read by a POSA. Patent claims must be interpreted through the lens of a POSA in view of the entire patent.

Declaration of Dr. Tim A. Williams
U.S. Patent No. 6,339,757

Accordingly, except where otherwise specified, in formulating my opinions, I have reviewed the claims of the '757 patent as I perceive a POSA would have understood them at the time of the earliest priority date (February 19, 1993) in light of the entire '757 patent specification.

26. Finally, I have been informed that claim construction is ultimately a question of law. Accordingly, I understand that a tribunal may choose to construe certain terms to provide clarity to the proceeding should any dispute arise between the parties over how a term should be construed. If the Board chooses to construe any term, I reserve my right to review and potentially modify any opinions below in view of such constructions.

C. My understanding of obviousness

27. 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 the application was filed. This means that, even if all the requirements of the claim cannot be found in a single prior art reference that would anticipate the claim, the claim can still be invalid.

28. To obtain a patent, a claimed invention must have been, as of the priority date, 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 to a POSA.

29. I understand that, to prove that prior art or a combination of prior art references renders a patent claim 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 claim appear in each of the asserted references; and (3) explain how the prior art references would have been combined in order to create the inventions claimed in the asserted claim.

30. I understand that certain objective indicia can be important evidence regarding whether a patent is obvious or nonobvious. Such indicia include: (1) commercial success of products covered by the patent claims; (2) a long-felt need for the invention; (3) failed attempts by others to make the invention; (4) copying of the invention by others in the field; (5) unexpected results achieved by the invention as compared to the closest prior art; (6) praise of the invention by the infringer or others in the field; (7) the taking of licenses under the patent by others; (8) expressions of surprise by experts and those skilled in the art at the making of the invention; and (9) the patentee proceeded contrary to the accepted wisdom of the prior art.

V. BACKGROUND OF THE TECHNOLOGY

31. The '757 patent is directed to encoding audio signals that have been converted from analog to digital format, and, in particular, a method for allegedly

decreasing the error in the encoding process by more efficiently allocating bits to the signal. EX1001, Abstract, 1:7-10. Before addressing the patent specification and claims, I first describe the state of such technology as of February 1993.

A. General background of digital audio signal encoding

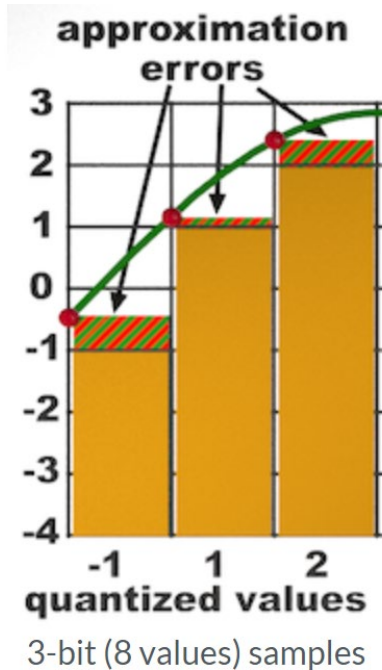
32. An analog audio signal is typically represented by a continuous waveform of amplitude values as a function of continuous or discrete time. Each amplitude represents the amplitude of the signal (e.g. voltage). EX1040, 27-29.

33. For easier storage and processing in computer systems, however, analog signals must be converted into a format that uses discrete numerical values for the amplitude of the signal at discrete time intervals rather than a continuous amplitude and time waveform. These converted signals are called “digital” signals. EX1040, 18.

34. When an analog signal is converted to digital form, time is broken up into segments called “sample intervals.” Additionally, the amplitudes of the analog waveform are mapped onto a finite number of values. EX1040, 18; *see also* EX1040, 102-05. The amplitude values are represented in digital form by a digital word comprised of a number of binary digits or bits (e.g., 1 or 0). The more binary digits that are included within the digital word, the more accurately the digital form can represent the amplitudes of the original analog waveform. The error caused by the conversion of the amplitudes from analog to digital is called “quantization

error.”

35. The figure below depicts the basic concept of amplitude values being quantized, which was well known by February 1993:



EX1033, 26.

36. The figure shows an analog signal (green) sampled at predetermined time intervals. As shown, each amplitude at a sampled time (red circles) must be rounded up or down to the nearest available digital value in a process known as “quantization.” EX1001, 1:43-45; EX1040, 18; *see also* EX1040, 102-05. The larger the numerical distance between an amplitude and its corresponding quantized value (shown as a hatched red and green rectangle in the figure above), the greater the error in the digital form or the encoded signal. EX1006, 1:33-39.

37. One way to reduce this error is to increase the number of values that a

digital word (integer) can contain. Doing this requires allocating more bits to the digital word. But when the total number of bits available in the system is limited, decisions have to be made regarding the allocation of bits. The process of deciding how to allocate the available bits is known as “bit allocation.”

38. The more bits that are allocated, the greater the number of values an integer may have and thus the lower the quantization error. For example, if the number of allocated bits is 2, the amplitude can only be expressed in four different values. But if the number of allocated bits is 8, the amplitude can be expressed in 256 different values, making it possible to accurately reproduce even subtle changes in amplitude. EX1006, 1:33-39; EX1008, 2:7-11; EX1009, 1:38-41.

39. The total number of bits used by a system, however, is limited by the available bit rate of the channel for communicating the audio signal. *See* EX1033, 26 (“The number of available values which can be assigned is determined by the number of bits (0s and 1s) used for each sample, also called bit depth or bit resolution.”). Higher bit-rate channels reproduce highly accurate digital signals but are slow to convey the desired information and costly. This is because transmitting a large number of bits creates “a traffic jam ... in the channels provided for transmitting the bits, in the memories provided for storing said bits and/or in the circuits for processing said bits, which increases the cost of the corresponding equipment.” EX1009, 1:41-46. And so, the challenge is to reduce quantization

error for low bit-rate signals or, said another way, to achieve the best possible performance given the available bits of the communication channel. EX1006, 1:17-25; EX1009, 1:46-55. The process of bit allocation is well studied since it is important for achieving the best performance for a given communication channel.

40. Fortunately, not all values within a waveform need to be quantized with great precision (or at all). This is because some frequencies, although present in the original signal, are not perceptible to the human ear and thus need not be reproduced in the encoded signal and others require less precision due to the characteristics of human hearing. By contrast, some portions of the waveform are more important in terms of human perception than others and, therefore, “deserve more bits in the coding process.” EX1014, 4:26-28. And so, one way to create an accurate digital signal using a bit-rate constrained channel is to allocate more bits to frequencies that matter and fewer bits to those that do not.

41. This method of allocating bits is based on “psychoacoustic criteria,” or “how an individual perceives sound.” EX1002, 194; EX1007, 1:32-40; EX1001, 1:57-67. One example of such a method is to process an audio signal by first transforming it into the frequency domain and dividing the resultant spectrum into multiple “bands,” each band containing a number of “spectral lines” whose amplitudes represent the magnitude of the frequencies within that particular band that make up the original signal. EX1031, 159; EX1032, II-198; EX1007, 1:9-31.

An example of these spectral lines is shown below:

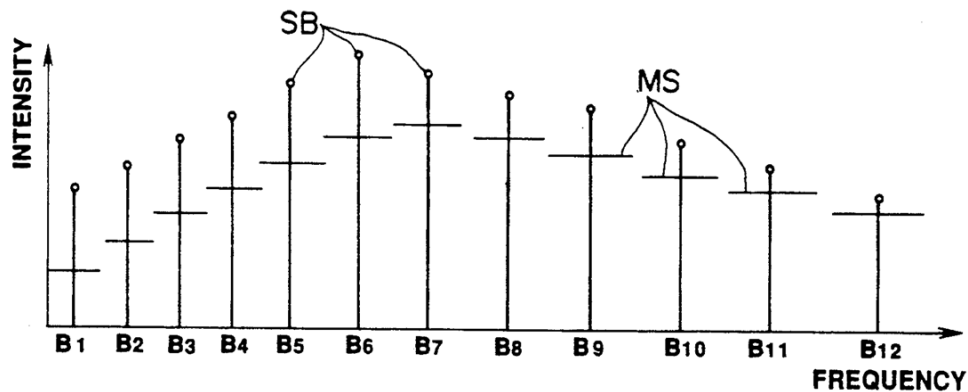


FIG.7

EX1007, FIG. 7.

42. The figure also demonstrates the concept of “masking,” which is when certain spectral lines are rendered inaudible by more dominant bands having spectral lines with greater magnitudes. EX1007, 5:65-68; 6:21-26. The horizontal lines labeled “MS” represent the threshold below which any signal or noise becomes inaudible due to the masking effect of louder signals. Notably, when the overall power of the spectrum increases, the masking spectrum also rises, showing that louder sounds can mask a wider range of frequencies around them.

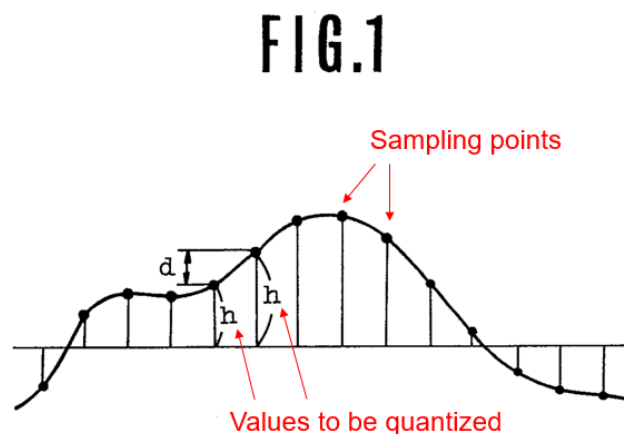
43. Rather than allocating bits evenly across each band, bits can be allocated based on how perceptible the frequencies within the bands are to the human ear. EX1031, 159; EX1005, 1:60-68. That way, the most perceptible bands are quantized with the greatest precision (i.e., allocated more bits), while masked bands are quantized with lower precision or not at all (i.e., allocated fewer bits).

EX1001, 3:14-26, FIG. 5; EX1031, 167. This type of bit allocation can be performed in one or more iterative passes—i.e., any leftover bits not allocated to a frequency band in the first pass are allocated in the next pass.

44. As explained below, all of these principles were well known in the art as of February 1993.

B. Quantization of signals for bit allocation was well known prior to the '757 patent.

45. Many references demonstrate that the quantization of signals was well known. For example, European Patent Application Publication No. 0145788A1 to Akagiri (“Akagiri’788”), titled “Method and Apparatus for Transmission of Digital Signal,” describes how to transmit audio signals by quantizing and coding. EX1012, 1:18-23. An example of Akagiri’788’s audio signal is depicted in Figure 1 below:



EX1012, FIG. 1 (annotated).

46. The waveform in Figure 1 is an analog audio signal that has been

“sequentially sampl[ed]” and depicts the amplitude of the signal at certain times.

EX1012, 6:14-16, 8:12-9:1. Akagiri’788 explains that these amplitude values are “quantized and encoded.”

47. Akagiri’788 depicts conversion of a waveform from an analog waveform (continuous amplitude and time) into a digital waveform (discrete amplitude and time). It was well known as of February 1993 that time waveforms and frequency spectra are related by the Fourier transform. Frequencies having non-zero coefficients are the frequencies that are contained in the time waveform.

48. In sub-band coders the information about the magnitude and phase contained in each of the spectral lines (bins) are transmitted between the transmitter and the receiver. Keeping the number of bits low that need to be transmitted is important in communication systems since these transmissions are expensive. Therefore, the process quantizing these magnitude and phase values for the spectral lines to result in as low a number of bits as possible with the greatest humanly perceived fidelity is the goal of the system.

49. For example, European Patent Application Publication No. 0525809A2 to Akagiri (“Akagiri’809”), titled “Digital Encoder with Dynamic Quantization Bit Allocation,” describes how to compress “a digital input signal derived from an analog signal to reduce the number of bits required to represent the analog signal with low quantizing noise.” EX1013, Abstract. Akagiri’809’s

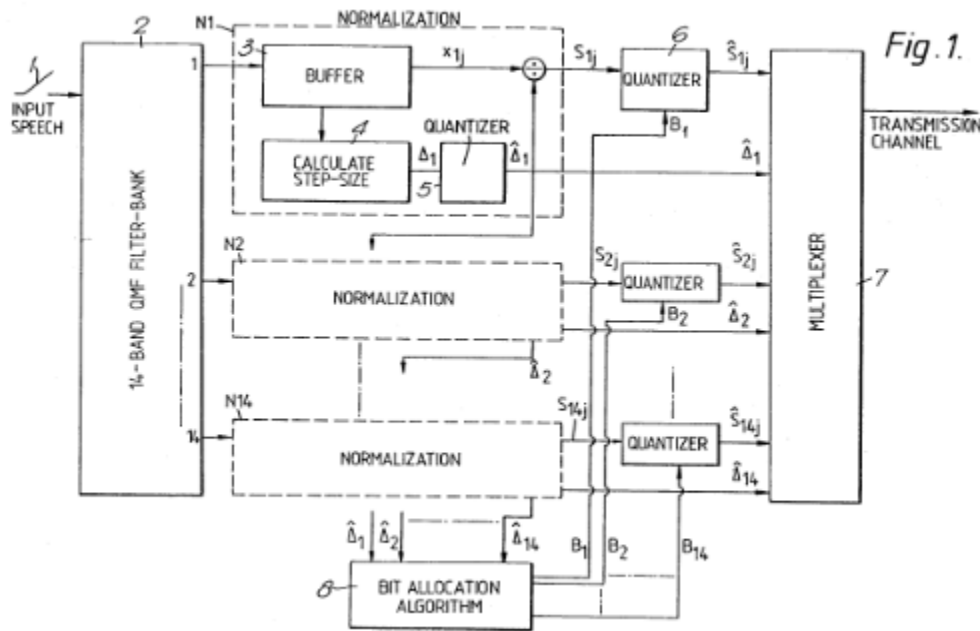
apparatus “receives [a] digital input signal and divides the digital input signal into a plurality of frequency ranges.” EX1013, 3:23-26. “[A] frame of digital signals representing the audio signal on the time axis is converted by an orthogonal transform into a block of spectral coefficients representing the audio signal on the frequency axis.” EX1013, 1:17-22. “[T]he spectral coefficients within each critical band are quantized using a number of quantizing bits allocated to each critical band.” EX1013, 8:28-31.

50. These are just a handful of examples. There are many others. *See, e.g.*, EX1015 (“Frequency Domain Speech Coding”); EX1016 (“Encoder/Decoder for Multidimensional Sound Fields”); EX1017 (“Adaptive Predictive Processing System”); EX1018 (“Multirate Digital Transmission Method and Device For Implementing Said Method”); EX1019 (“Adaptive Transform Coder Having Minimal Bit Allocation Processing”); EX1021 (“Adaptive Bit Assignment Transform Coding According To Power Distribution Of Transform Coefficients”); EX1022 (“Process for Transmitting Audio Signals”); EX1023 (“Error Detection Method For Sub-Band Coding”); EX1025 (“Transform Coding of Audio Signals at 64 KBIT/S”); EX1026 (“Predictive SVD-Transform Coding of Speech With Adaptive Vector Quantization”); EX1027 (“An Optimal Bit Algorithm for Sub-Band Coding”); EX1028 (“Efficient Bit Allocation for an Arbitrary Set of Quantizers”); EX1029 (“Quantization and Bit Allocation in Speech Processing”);

EX1041 (“Method For Transmitting Audio Signals”).

C. Dividing a frequency spectrum into frequency bands and assigning bits to those bands was well known prior to the '757 patent.

51. Dividing an audio signal into frequency bands (sometimes called “sub-bands”) and assigning bits to each band was also well known. For example, Koh, titled “Frequency Domain Speech Coding,” describes techniques related to an “[a]daptive bit allocation to the channels of a sub-band coder.” EX1015, Abstract.



EX1015, FIG. 1.

52. Koh’s “technique involves splitting the broadband signal into two or more frequency bands and encoding each band separately.” EX1015, 1:30-32. In Figure 1 above, filter bank 2 divides a full band spectrum of an input signal into fourteen sub-bands, and quantizer 6 quantizes each sub-band based on the bit

allocation determined by the bit allocation algorithm 8. EX1015, 2:66-3:34.

53. Akagiri'809 also discloses sub-bands obtained by dividing the higher frequency critical bands. EX1013, 8:28-34.

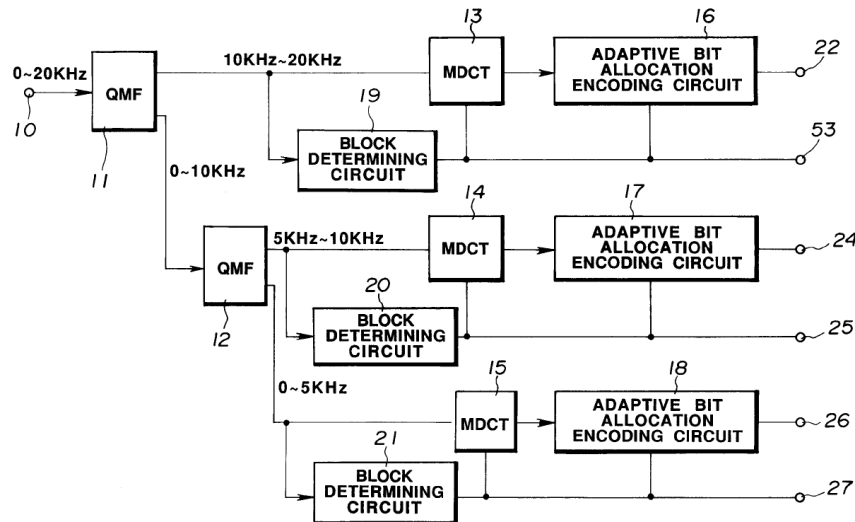


FIG.1

EX1013, FIG. 1.

54. Akagiri'809 explains how “a digital input signal is divided in frequency into a plurality of frequency ranges by a filter.” EX1013, 6:31-33. In Figure 1, Akagiri'809 also explains that “a quadrature mirror filter (QMF) is used to divide the spectrum of the input signal into a 0 Hz to 10 kHz frequency range and into a 10 to 20 kHz frequency range” and “[t]he 0 Hz to 10 kHz frequency range is further divided” “into a 0 Hz to 5 kHz frequency range and a 5 to 10 kHz frequency range.” EX1013, 7:27-39.

55. Likewise, Davis discloses techniques related to “invention relates to subband encoding and decoding of signals representing such sound fields.”

EX1016, Abstract.

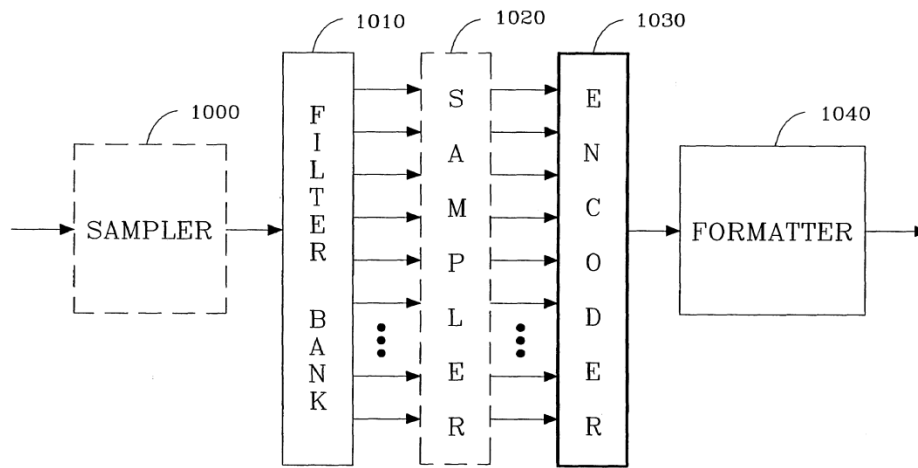


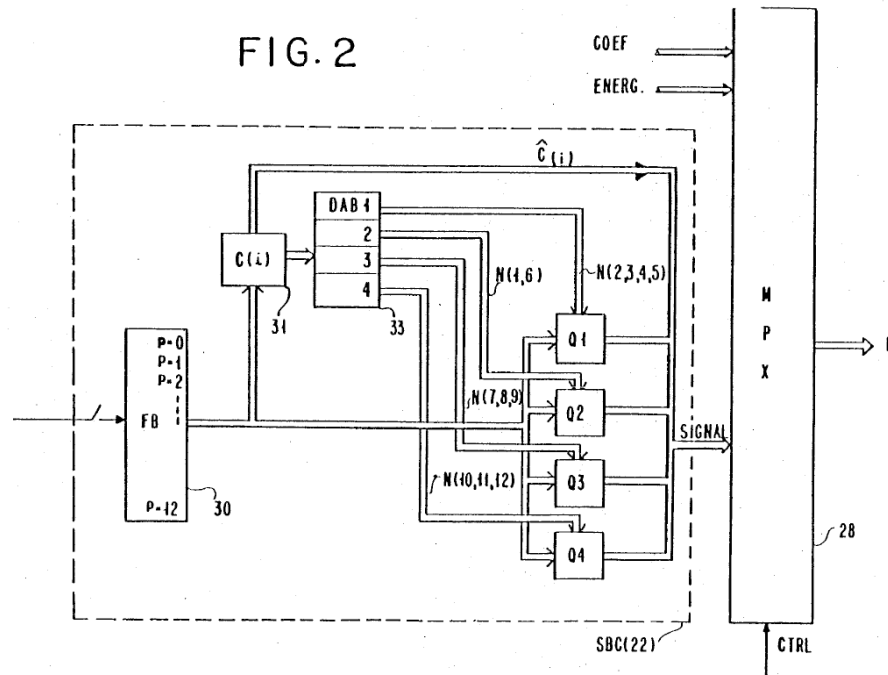
FIG. 4

EX1016, FIG. 4.

56. Davis teaches that “[s]ignal recording, transmitting, or reproducing techniques which divide the useful signal bandwidth into frequency bands with bandwidths approximating the ear’s critical bands can better exploit psychoacoustic effects than wider band techniques.” EX1016, 4:27-29. In Figure 4, as an example of the “[t]ypical single-channel subband encoding,” Davis explains that “Filter Bank 1010” “split[s] the input signal into 25 subbands” to “convert[] the subband information into quantized code words.” EX1016, 8:24-26.

57. Honda also teaches the technique which splits “speech signal frequency bandwidth” “into several sub-bands.” EX1018, Abstract. Honda explains the sub-band coding, mentioning that “it should be understood that the

speech signal frequency bandwidth is split into ‘p’ sub-bands,” and “[t]he samples provided in each sub-band are quantized (or requantized) with a variable precision based on the relative energy of the signal contained in said sub-band.” EX1018, 4:62-5:2.



EX1018, FIG. 2.

58. Mazor also discloses “quantizers designated for individual subbands.” EX1020, Abstract. Mazor teaches that its quantizer divides “the spectrum” “into twenty subbands” “to minimize distortion due to quantization error.” EX1020, 2:65-66, 4:10-12.

59. Again, these are just a few examples. There are many others. *See, e.g.*, EX1017 (“Adaptive Predictive Processing System”); EX1020 (“Dynamic Bit

Allocation Subband Excited Transform Coding Method And Apparatus”); EX1021 (“Adaptive Bit Assignment Transform Coding According To Power Distribution Of Transform Coefficients”); EX1027 (“An Optimal Bit Algorithm for Sub-Band Coding”); EX1028 (“Efficient Bit Allocation for an Arbitrary Set of Quantizers”).

D. Assigning bits based on psychoacoustic factors was a well-known technique prior to the '757 patent.

60. Still further, assigning bits based on psychoacoustic criteria was also well known in the art. For example, Akagiri’809 explains that “us[ing] sub bands that take advantage of the psychoacoustic characteristics” produces a more accurate digital signal.” EX1013, 1:50-2:9. Akagiri’809 further discloses a method of encoding “spectral coefficients” “within each critical band” identified based on psychoacoustic characteristics “using a number of quantizing bits allocated to each critical band.” EX1013, 8:28-31.

61. Similarly, Davis allocates bits based on “psychoacoustic effects” associated with the “human ear.” EX1016, 4:14-17. According to Davis, “the frequency-resolving and direction-discerning ability of the human ear varies according to the bandwidth of these filters throughout the audio spectrum,” and “[t]he effective bandwidth of such an auditory filter is referred to as a ‘critical band.’” EX1016, 4:20-22. Davis further teaches that noise can be reduced by allocating bits using this critical band “without any audible degradation by 15 exploiting various psychoacoustic effects.” EX1016, 4:14-15.

62. I note that bit allocation based on psychoacoustic criteria is sometimes described in the context of masking. As explained above, masking refers to a dominant frequency or band rendering inaudible an adjacent frequency or band within the signal. When allocating bits based on psychoacoustic criteria, fewer bits are allocated to frequency bands that are masked. Accordingly, the two phrases are often used interchangeably. In fact, the '757 patent uses both terms without distinguishing between them. EX1001, 1:56-60 (“The first category consists of bit allocation methods applying the psychoacoustic phenomena of simultaneous masking and threshold in quiet to derive the masking threshold.”).

63. Many prior art references discuss methods for bit allocation based on masking effects. EX1031; EX 1016. For example, “Low bit-rate coding of high-quality audio signals: An introduction to the MASCAM System” (“Theile”) introduces multiple coding methods that apply “masking effects.” Theile teaches that its method “enables a reduction in the redundancy of the audio signal and also a matching of the quantization to the thresholds of perception of quantizing errors” because “[o]nly those values of amplitude and phase that are relevant with respect to the masking effects of the human ear are quantized.” EX1031, 159.

64. Once again, these are examples—there are others as well. *See, e.g.*, EX1013 (“Digital Encoder With Dynamic Quantization Bit Allocation”), 6:37-41; EX1015 (“Secure Digital Speech Communication”); EX1024 (“a 128 kb/s Hi-Fi

Audio CODEC Based on Adaptive Transform Coding with Adaptive Block Size MDCT”); EX1025 (“Mahieux NPL”); EX1028 (“Efficient Bit Allocation For An Arbitrary Set of Quantizers”).

E. Assigning bits using a two-step bit allocation process was a well-known technique prior to the '757 patent.

65. Finally, allocating bits using two distinct passes was also well known. For example, Akagiri’809 discloses a technique for assigning two types of bits: bits with fixed values and bits based on signal level.

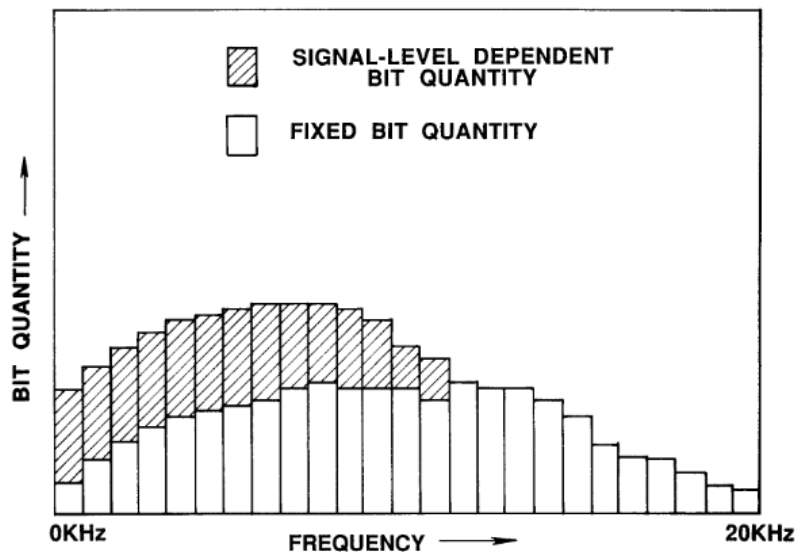


FIG.5

EX1013, FIG. 5.

66. In Figure 5, the non-hatched area represents bits allocated according to a selected predetermined bit allocation pattern, and the hatched area represents the quantity of bits allocated depending on signal energy in the critical bands.

EX1013, 10:44-49. By allocating bits based on two different criteria, the method

achieves a higher degree of accuracy and precision.

67. Vössing more explicitly teaches a two-step bit allocation process. Even “[a]fter a preliminary approximative quantification,” Vössing notes, “there remains the problem of distributing the still available bits of a data block among the spectrum components.” EX1041, Abstract. To address that problem, Vössing performs “a process of precise quantification.” EX1041, Abstract.

68. At “[t]he first step in bit allocation,” Vössing teaches that a “specified number of bits” are allocated to “the absolute magnitude maximum, the magnitude maximum in frequency groups of the short-time spectrum, the addresses of the magnitude maximum and the dynamics of the frequency groups.” EX1041, 0005. Vössing calls this process as “an approximative quantification” and explains that the quantification “is dimensioned so that the number of bits available for coding is sufficient for an initial assignment, even for short-time spectra with many magnitudes and phases to be transmitted.” EX1041, 0006.

69. After the first step, “there is a residual number of bits available for distribution.” EX1041, 0007. This is because “bits can only be distributed as integers,” while the calculated number of bits at the first step is “generally not” represented by integers. EX1041, 0007. Therefore, to make effective use of these remaining bits, Vössing distributes them “by rounding and transmitting the remainder according to the absolute magnitudes of the magnitudes of the

spectrum.” EX1041, 0009.

70. Against this technological backdrop, I now turn to the '757 patent.

VI. OVERVIEW OF THE '757 PATENT

71. The '757 patent covers these same features already known in the art. Titled a “Bit Allocation Method for Digital Audio Signals,” the patent describes a two-step bit allocation method that is “adapted to the human auditory system” and has “a low level of complexity so that high audio quality can be achieved while at the same time meeting the low cost target.” EX1001, 2:40-45.

72. The patent states that “[t]he greatest difficulty in the design of a dynamic bit allocation procedure lies in balancing good audio quality while at the same time, keeping the procedure as simple as possible.” EX1001, 2:13-16. The patent categorizes conventional bit allocation schemes into two categories: “[t]he first category consists of bit allocation methods applying the psychoacoustic phenomena of simultaneous masking and threshold in quiet to derive the masking threshold,” while “[t]he second category offers a more simplified method of allocating according to the signal statistics.” EX1001, 1:57-60, 2:1-2.

73. According to the patent, bit allocation methods in the first category are “able to produce good sound quality,” but “a great deal of complexity is incurred” given that they “require full DSP (either fixed point or floating point in some cases) power in order to perform the bit allocation.” EX1001, 2:16-19. By

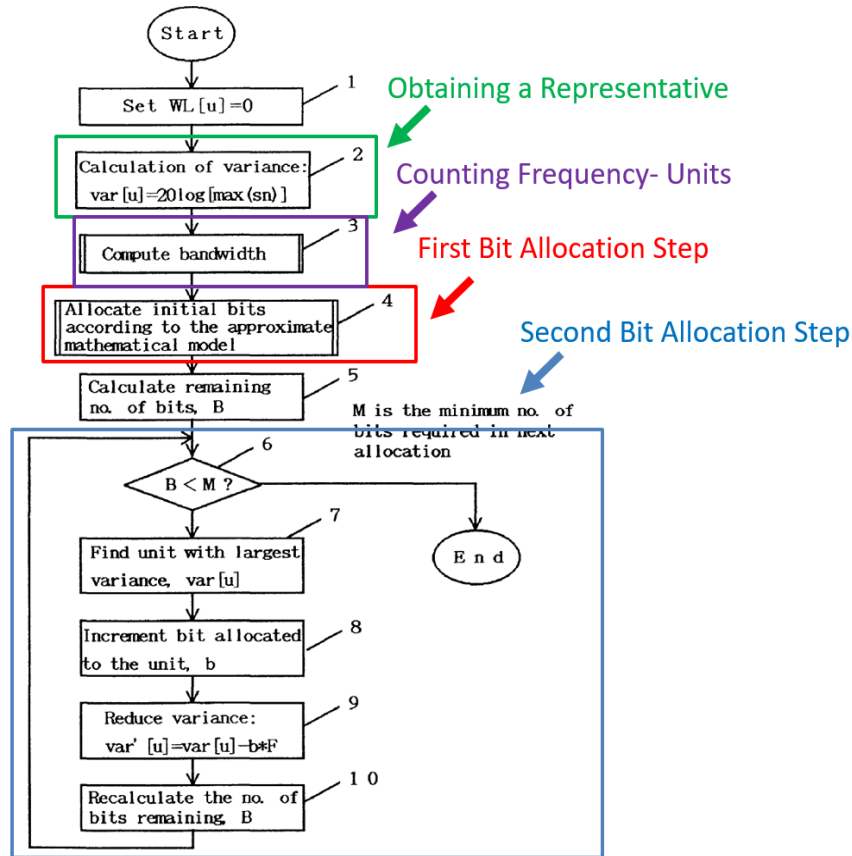
contrast, bit allocation methods in the second category, “while less complex,” are “unable to attain the quality of the perceptual based bit allocation as [their] design[s] [are] concentrated purely on the source which is the audio signal and not on the final receiver, the human auditory system.” EX1001, 2:22-26.

74. To address these purported challenges, the '757 patent describes “a dynamic bit allocation method which produces low hardware complexity and high quality audio, by combining psychoacoustic criteria and human hearing threshold.” EX1001, 2:60-63.

A. The '757 patent's bit allocation scheme

75. The '757 patent's bit allocation method is shown in Figure 1, reproduced below:

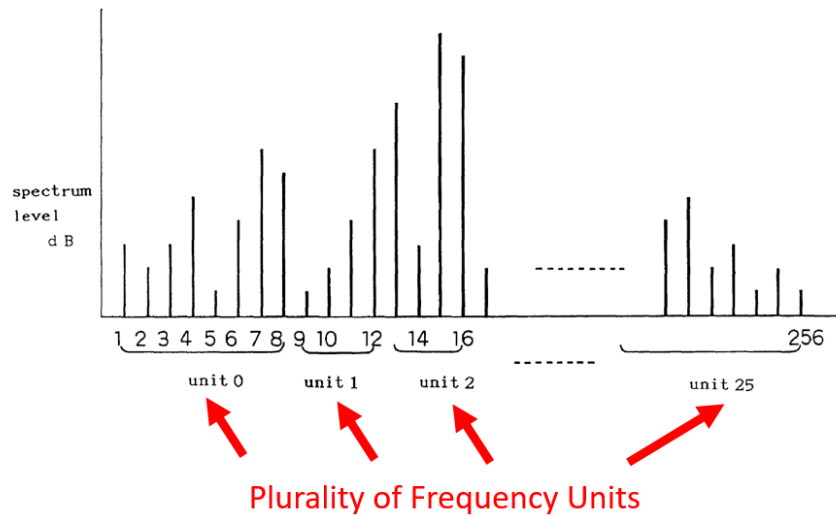
F i g . 1



EX1001, FIG. 1 (annotated).

76. As a preliminary matter, a sampled signal is first transformed into the frequency domain and divided into multiple frequency bands or “units.” EX1001, 1:29-45, 4:21-24. This step is not shown in Figure 1, but the frequency-units are depicted in Figure 6:

F i g . 6



EX1001, FIG. 6 (annotated).

77. The figure shows a signal divided into twenty-five frequency units, each containing 256 spectral lines or frequency components representing the frequency content of the original analog signal. EX1001, 4:21-28; *see also* EX1001, FIG. 7. The spectral lines are the bins of the Discrete Cosine Transform (DCT) of the discrete version of the original analog signal. As the patent explains, “[t]he transformed spectral samples are grouped into units and in each unit, a scale factor is derived from the peak values of the samples in the unit, 8. These units are non-uniform frequency intervals with a finer resolution in the low frequencies and coarser resolution in the higher frequencies.” EX1001, 1:37-43.

78. The purpose of the method depicted in Figure 1 is to determine how many bits should be allocated to each frequency-unit. The number of bits as a

function of unit is represented by “WL[u]”—the goal of the patent is to calculate WL[u] (i.e., to populate the third column in the table below):

F i g . 7

unit number	spectral line number	bit allocation data
unit 0	8	WL (0)
unit 1	4	WL (1)
unit 2	4	WL (2)
unit 3	4	WL (3)
unit 24	10	WL (24)
unit 25	10	WL (25)

EX1001, FIG. 7.

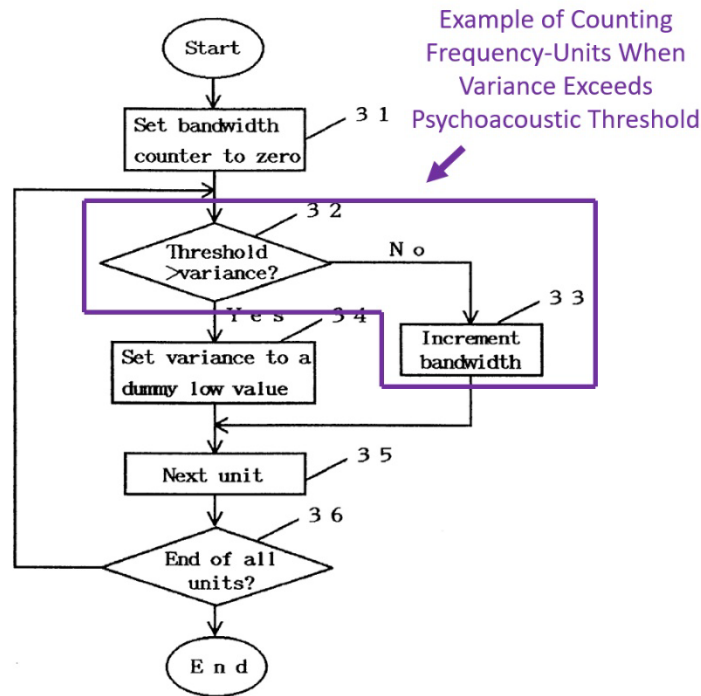
79. To that end, the method in Figure 1 begins at step 1, during which “the quantizations or bit allocations for each frequency interval or unit (WL[u]) is set to zero.” EX1001, 3:48-49. This step initializes the WL[u] in preparation for later steps.

80. At step 2, for each frequency-unit, a so-called “representative” is calculated. The ’757 patent does not attribute a particular meaning to that term, but it is clear from context that a frequency-unit’s representative is simply a characteristic of the unit that is representative of the signal. EX1001, 2:46-57, 3:1-5; EX1002, 254. Notably, embodiments described in the patent use a so-called

“variance” as the representative. EX1001, 3:50-52, FIG. 1. According to the patent, the variance can be “calculated from the maximum spectral amplitude” within each unit (logarithmically scaled, as shown in step 2 of Figure 1) or calculated “from an average of the spectral components” within each unit. EX1001, 3:50-59. Thus, the word “variance” is not used in the traditional statistical sense and is instead being used in a non-standard way, making it difficult to understand exactly what type of calculation must be performed to determine the variance. Nevertheless, as I explain below, the prior art cited here mirrors the ’757 patent’s own disclosure.

81. At step 3, “[t]he bandwidth of transmission is computed.” EX1001, 3:59-60. It is important to note that this process does not calculate bandwidth in the traditional sense (as a range of frequencies). Instead, it counts the number of units that exceed a psychoacoustic threshold, which effectively determines the “useful” bandwidth for bit allocation purposes. The patent describes this step in the context of Figure 2:

F i g . 2



EX1001, FIG. 2 (annotated).

82. As shown, each unit's variance is "compared against the threshold" (step 32) that is "obtained from listening tests." EX1001, 4:29-48. If a variance exceeds the threshold, a "bandwidth counter is incremented" such that the unit will have bits allocated to it (step 33).¹ EX1001, 4:41-42. This process is repeated for

¹ I note that the portion of the specification describing Figure 2 refers to these two steps as corresponding to numerals 34 and 35, respectively. EX1001, 4:39-42. That appears to be a typographical error—as shown in Figure 2, those steps correspond to numerals 32 and 33.

all units, which ultimately “eliminate[s] units that are irrelevant according to psychoacoustic criteria.” EX1001, 4:45-48.

83. Next, turning back to Figure 1, step 4 describes the first-bit allocation step: “the initial number of bits are allocated according to the approximate mathematical model.” EX1001, 3:60-62. The patent does not limit the type of model that can be used but provides just a single example of such a model in Equation 1, reproduced below:

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

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

EX1001, 4:62-65, FIG. 3.

84. As shown above, the value “F” is described as a “psychoacoustic weighting factor.” The patent provides no specific method for determining F. Rather, F is a constant adjusted to whatever value produces the least amount of noise for a given signal. As the patent explains:

The mathematical model has the companding effect of allocating bits such that the error spectrum produced will follow the signal spectrum more closely. This is achieved by setting the value of F to be between 8.5 and 10. When F=6, a flat error spectrum is obtained and this was found to produce high frequency artifacts for a number of test sequences at the bit rate of operation.

FIG. 4 illustrates the spectral plot of a frame of violin signals, the error spectrum when F=9.03 and the error spectrum when F=6. As can be observed from the plot, the F=9.03 error spectrum shows a higher value of error in the low frequencies but these errors are not noticeable as they are masked by a much stronger signal in the band.

EX1001, 5:11-24. In other words, the value of F is adjusted to obtain the desired characteristics, not a theoretically determined value.

85. Finally, once again returning to Figure 1, steps 6-10 describe the second bit allocation step: “[t]he remaining bits, B are iteratively allocated to the units” “until the number of bits remaining is less than the minimum number of bits required in the next allocation.” EX1001, 3:63-67. This second step is designed to “increment the bit allocations in the unit by one or two bits (b), $WL'[u]=WL[u]+b$,” to “the unit with the largest variance” “until the number of bits remaining is less than the minimum number of bits required in the next allocation.” EX1001, 3:63-11. To prevent additional bits from continuing to be allocated to only one unit, the '757 patent “[r]educe[s] the variance of the unit by

$\text{var}'[u] = \text{var}[u] - b * F.$ " EX1001, 4:11-13. The process is then repeated until all bits are allocated.

B. The patent claims

86. The '757 patent contains 21 claims, but only claims 1-17 are challenged. Independent claim 1 is representative and reproduced below (color coded to match the annotated steps of Figure 1 above):

[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-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] a second bit allocation step for determining further bit allocation for each of said available frequency-units based on psychoacoustic criteria.

EX1001, 5:41-61.

87. The patent's dependent claims add limited specificity regarding: (1) the mathematical model used in the first bit allocation step (claim 2), (2) the type of representative used (claims 3-5, 7-9, 11-15, 17), (3) the second bit allocation step (claim 6), and (4) how frequency-units are obtained (claim 16) and counted (claim 10). EX1001, 5:62-6:43.

C. The prosecution history of the '757 patent

88. I understand that, during prosecution, the '757 patent was rejected several times over various prior art references. Specifically, in a series of office actions, the claims were rejected under 35 U.S.C. § 112 and as obvious over U.S. Patent No. 5,105,463 to Veldhuis et al.² (Veldhuis'463, EX1011) and U.S. Patent No. 5,151,941 to Nishiguchi et al. (Nishiguchi, EX1030). In response, the Applicant amended the claims and added claim 5, which is substantially similar to issued claim 1 (except that it omits the step of "dividing at least one frequency interval into a plurality of frequency-units"). The Applicant argued that:

² I understand that Veldhuis'463 is a divisional of the Veldhuis prior art reference relied upon in Grounds 3-4 of the petition.

New claim 5, which replaced cancelled claim 1, provides for two bit allocation steps based on psychoacoustic criteria for all frequency ranges and counts the number of available frequency-units. In comparison, Veldhuis et al. teach the use of a set number of quantization levels that are predetermined for the lower half of the frequency ranges, col. 7, line 68 to col. 8, line 1. Veldhuis et al. disclose a bit allocation for the frequency ranges in the upper half, col. 10, lines 8-18. Veldhuis et al. do not teach the counting of available frequency-units. The counted number of frequency-units in Applicants' method is not synonymous with Veldhuis' use of bandwidth as, implied in paragraph 6(c) on page 5 of the Office Action. Furthermore, the mathematical model relied upon in Veldhuis et al. does not utilize available frequency-units. Consequently, since there is at least one element present in Applicants' application that is not present in Veldhuis et al., this rejection is overcome as the remainder of the claims depend from a patentable claim.

EX1002, 157-158.

89. I disagree with these statements. As I explain in greater detail below in Sections X.D.4-5, Veldhuis teaches, or at least renders obvious, counting the available frequency units and using this information in a first bit allocation step.³

³ As I also explain below, to the extent Veldhuis alone does not teach or render obvious these limitations, Veldhuis in combination with Galand does. See Sections X.D.4-5.

90. Ultimately, the Applicant appealed the Examiner's rejections to the Board of Patent Appeals and Interferences (BPAI). *See* EX1002, 188-203 (Appeal Brief), 219-235 (Reply Brief). Among other things, the Applicant argued: (1) "[t]here is no specific recited limitation that prevents each of the frequency-units from being an available frequency-unit," EX1002, 194; (2) "[t]he consequence of not being an available frequency-unit is that no additional bits are assigned or allocated to that frequency-unit," EX1002, 195; (3) "Applicants use the term 'variance' in the specification to define a change within each frequency-unit," EX1002, 197; (4) Veldhuis'463 "does discuss the allo[c]ation [of bits] for the upper half of the frequency ranges" (i.e., the portion relied upon in Grounds 3-4), EX1002, 198, 200-201; and (5) "there is no mention of a bandwidth counter" in the claims, EX1002, 225-226.

91. The BPAI made its own findings regarding many aspects of Veldhuis'463's disclosure and agreed with the Examiner's application of Veldhuis'463 in many respects. For example, the BPAI found that: (1) buffering the signal in frames "is considered either inherent or obvious" in view of Veldhuis'463, EX1002, 253; (2) Veldhuis'463 discloses "decomposing the digital signals into spectral components" and 'a step for dividing a least one frequency interval into a plurality of frequency-units,'" EX1002, 253 (citing EX1011, 5:58-64, 6:40-46); (3) the "claim limitation of 'at least one frequency interval' reads on

the frequency interval corresponding to the upper subbands ($13 \leq p \leq 26$)” in Veldhuis’463, EX1002, 253; (4) the “set of ‘characteristic parameters $G(p;m)$ ’ in [Veldhuis’463] is the ‘representative of each frequency-unit obtained,’” EX1002, 254; (5) Veldhuis’463’s “thresholds are ‘based on psychoacoustic criteria,’” EX1002, 255 (citing EX1011, 11:31-44); (6) the “number of blocks $(p;m)$ in the allocation window having $C(p;m)=1$ ’ is the count of available frequency-units,” EX1002, 256; and (7) the phrase “based on psychoacoustic criteria” can be “interpreted broadly as very indirectly based on some psychoacoustic criterion,” EX1002, 257.

92. But, ultimately, the BPAI reversed the Examiner’s rejection on three grounds. First, the Board stated that Veldhuis’463 does not disclose “a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria” and that “neither Veldhuis nor Nishiguchi discloses counting the number of available frequency-units.” EX1002, 258. In my view, however, and as I explain in more detail below, a POSA would have at least found it obvious to count the available frequency-units in light of Veldhuis’s disclosure, as further confirmed by Galand. *See* Section X.D.4.

93. Second, the Board stated that Veldhuis’463 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.” EX1002, 258, 260, 262. However, as I explain in more detail below, Veldhuis in combination with Galand teaches using a “mathematical model” for determining the initial bit allocation, as claimed. *See* Sections X.C, X.D.5.

94. Third, the Board stated that Veldhuis’463 does not disclose “a second bit allocation step for determining further bit allocation for each of said available frequency-units based on psychoacoustic criteria.” EX1002, 173. In my view, however, and as I explain in more detail below, Veldhuis assigns bits only to those blocks that can be heard, thus taking into account psychoacoustic criteria. *See* Section X.D.6. Further, Veldhuis’s second bit allocation is implemented in the same way as described in the ’757 patent, and a POSA would have understood that Veldhuis’s D value does in fact reflect psychoacoustic criteria. *See* Section X.D.6.

95. Accordingly, even assuming that any of the BPAI’s adverse findings apply to the Veldhuis reference on which I rely here, I disagree with those findings. In my view, as explained below, Veldhuis, in combination with the other references cited here, renders claims 1-17 of the ’757 patent obvious. *See* Sections X-XI.

VII. CLAIM CONSTRUCTION

96. As shown above, the limitations recited in claim 1 of the ’757 include the phrase “step for.” I note, though, that neither the Applicant nor the BPAI

treated the limitations as step-plus-function terms.

97. To the extent the Board determines that the limitations are written in step-plus-function format, I identify below the claimed functions and disclosed acts. I note, however, that a number of claim terms are not sufficiently defined or supported by the patent specification (e.g., “representative,” “variance,” “counting,” “mathematical model,” etc.), and so the disclosure below is my best attempt to identify the relevant acts.

Claim Term	Function	Corresponding Acts
“a step for dividing at least one frequency interval into a plurality of frequency-units”	Dividing at least one frequency interval into a plurality of frequency-units.	Grouping transformed spectral samples into units, and equivalents thereof. EX1001, 1:37-40, 4:21-24, FIG. 6.
“a step for obtaining a representative of each frequency-unit obtained”	Obtaining a representative of each frequency-unit obtained.	Calculating a variance from the maximum spectral sample amplitude or from an average of the spectral components within the unit, or determining the maximum spectral amplitude, and

Declaration of Dr. Tim A. Williams
U.S. Patent No. 6,339,757

Claim Term	Function	Corresponding Acts
		equivalents thereof. EX1001, 3:50-59, FIG. 1 (numeral 2).
“a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria”	Counting a number of available frequency-units by using the representative, based on psychoacoustic criteria.	Incrementing a bandwidth counter if a variance is above a human hearing threshold, and equivalents thereof. EX1001, 4:29-48, 4:51-53, FIG. 1 (numeral 3), FIG. 2 (numerals 32-33), FIG. 3 (numeral 41).
“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	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.	Allocating bits based on Equation 1 of the '757 patent, and equivalents thereof. EX1001, 4:49-5:11, FIG. 1 (numeral 4), FIG. 3 (element 43).

Claim Term	Function	Corresponding Acts
frequency-units”		
“a second bit allocation step for determining further bit allocation for each of said available frequency-units based on psychoacoustic criteria”	Determining further bit allocation for each of said available frequency-units based on psychoacoustic criteria.	Iteratively allocating bits as described at 3:63-4:18 and FIG. 1 (numerals 6-10), and equivalents thereof.

98. As explained below, whether the terms are step-plus-function is immaterial because the cited art renders the claims obvious under both the step-plus-function constructions and the terms’ plain-and-ordinary meanings.

VIII. GROUND 1: CLAIMS 1-3, 6-7, 10-11, 14, AND 16 OF THE ’757 PATENT ARE OBVIOUS OVER THE COMBINATION OF MAHIEUX AND FIELDER.

99. As I explain below, it is my opinion that claims 1-3, 6-7, 10-11, 14, and 16 of the ’757 patent are obvious over the combination of Mahieux and Fielder.

A. Overview of Mahieux

100. U.S. Patent No. 5,185,800 (“Mahieux”) describes a bit allocation device for transformed digital audio-broadcasting signals with adaptive quantization based on psychoacoustic criteria. As I explain in further detail below,

the system uses a time-frequency transformation to convert audio signals into the frequency domain. It then applies psychoacoustic masking principles to determine which spectral components can be quantized more coarsely or eliminated entirely without perceptible loss of audio quality. In short, Mahieux discloses techniques for adaptively allocating bits to different frequency bands based on their perceptual importance, as well as methods for coding the allocation information efficiently.

101. More specifically, Mahieux describes “compressing a digital audio signal” by “allocating a specific number of bits for the expression of the coefficients of each frequency band of a transformed digital audio signal.”

EX1005, Title, Abstract. Mahieux’s bit allocation method “takes account of psychoauditive criteria” to quantize only the most dominant frequencies within the signal. EX1005, 3:1-3.

102. Mahieux’s method is depicted in the context of Figure 1 below:

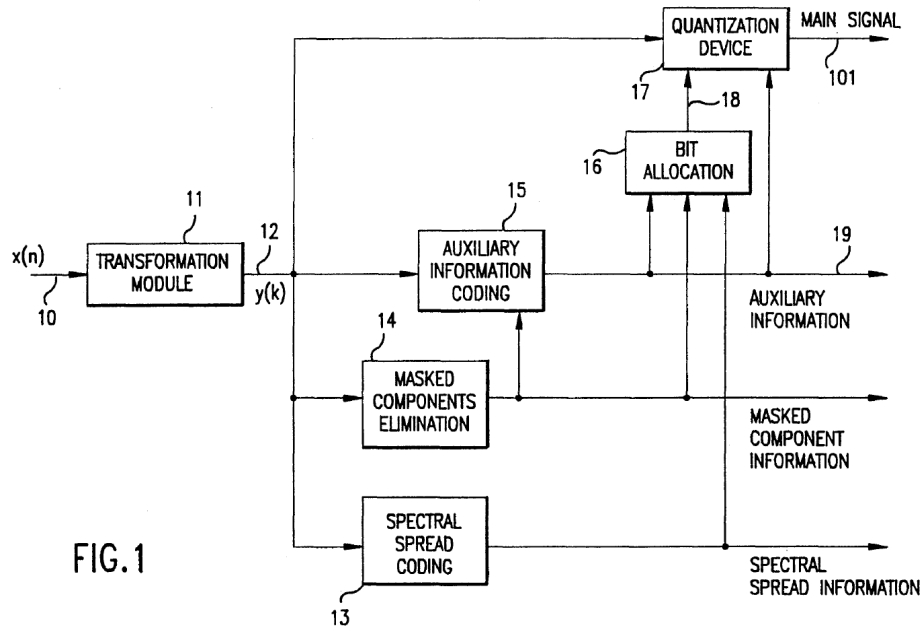


FIG.1

EX1005, FIG. 1.

103. First, an input signal ($x(n)$ 10) is sampled, decomposed into temporal blocks, and transformed by “transformation module” 11 to a frequency-domain representation. EX1005, 1:46-53, 2:13-34, 3:17-32, 5:14-36, 5:63-66. The frequency spectrum is then divided into a number of “bands” (shown in Figure 5 below as B_1 , B_2 , and B_3), each having one or more “spectral components” (51_1 , 51_2 , 51_3):

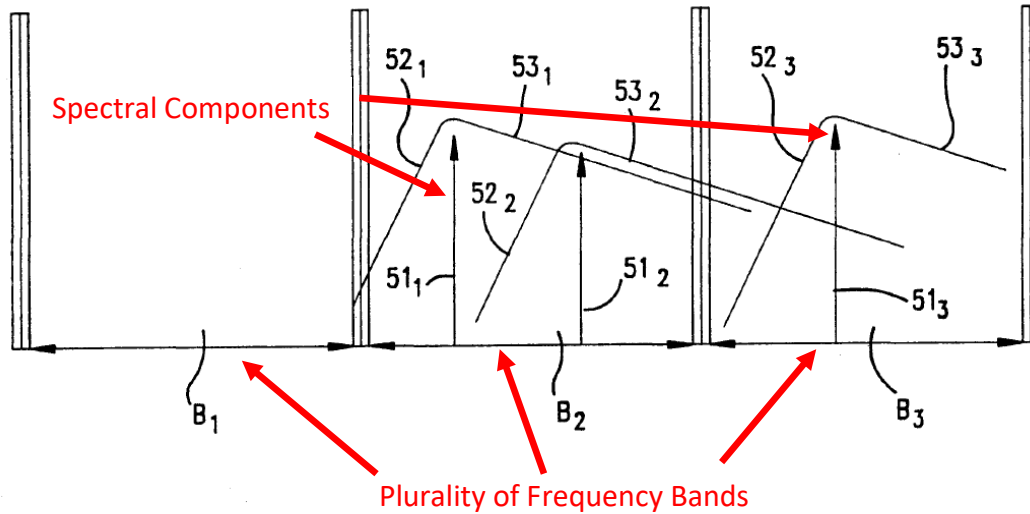


FIG.5

EX1005, FIG. 5 (annotated).

EX1005, 3:17-32.

104. Second, for each band, “transform coefficients” $y(k)$ are calculated. EX1005, 1:20-37, 2:13-34, 5:20-36. The coefficients represent the magnitudes of the frequency bands and thus the spectral energy of the audio signal at various frequencies—the square of the coefficients is proportional to the power. EX1005, 3:17-32, 5:63-65; *see also* EX1005 1:46-68, 2:13-34, 3:36-40, 6:40-46, 12:57-63. Mahieux explains that the coefficients are calculated by the following equation, where N is the size of the transform block, $h(n)$ is the block weighting window, and m is the block transform number:

$$y(m, k) = \sum_{n=0}^{N-1} x_m(n) \cdot h(N-1-n) \cdot \cos(2\pi(k + \frac{1}{2})(n + n_0)/N)$$

$$k = 0, \dots, N-1.$$

EX1005, 1:30-37; *see also* EX1005, 5:20-37.

105. Third, Mahieux considers spectral components “as a function of a psychoauditive criteria.” EX1005, 1:45-68, 2:9-12, 3:1-3, 3:17-52; *see also* EX1005, 3:33-49, 5:66-68, 9:48-59. “If a frequency band is entirely masked”—i.e., rendered inaudible by adjacent bands—Mahieux explains that “it is not necessary” to allocate bits to the band. EX1005 10:45-49; *see also* EX1005, 4:35-42, 5:66-6:6, 6:63-65, 7:45-8:6, 12:57-63. Such bands are not considered during the bit allocation process (in block 14 in Figure 1 above) and thus not counted. *See* EX1005, 6:30-33.

106. Fourth, Mahieux assigns bits to the counted bands (in block 16 of Figure 1). EX1005, 3:43-49, 6:30-46, 7:3-8:9, 11:33-12:64. Specifically, Mahieux allocates bits to frequency bands having “non-zero coefficients” according to the mathematical model depicted in Figure 7. EX1005, 11:33-38, 11:44-62, 12:21-29.

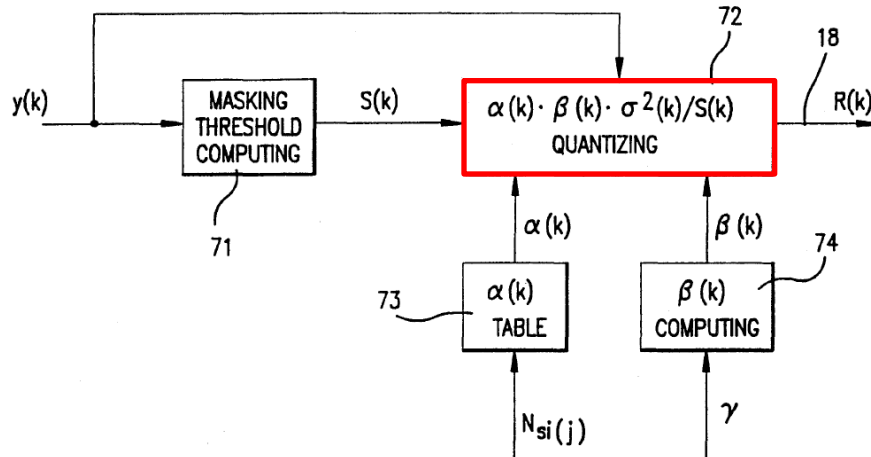


FIG.7

EX1005, FIG. 7, (annotated).

107. The “allocation of the bits” is performed by module 72, which quantizes “the ratio $\alpha(k) \cdot \beta(k) \cdot \sigma^2(k) / S(k)$.” EX1005, 12:3-11. This equation, which I explain in greater detail in Section VIII.D.5 below, provides a mathematical model for determining the number of bits to assign to a particular frequency band. EX1005, 12:12-56. The signal is then quantized by “quantizer” 17 to produce a digitally encoded signal. *See* EX1005, 6:49-58.

108. Mahieux’s method accounts for “psychoauditive criterion” to “enable a major reduction in the bit rate while, at the same time, preserving the quality of the starting signal to the maximum extent.” EX1005, 2:61-68.

B. Overview of Fielder

109. U.S. Patent No. 5,109,417 (“Fielder”) discloses a low bit rate

transform coder, decoder, and encoder/decoder for high-quality audio signals. As I explain in further detail below, the system applies a modified discrete cosine transform to convert time-domain audio samples into frequency-domain coefficients. Fielder then uses adaptive bit allocation to assign more bits to perceptually important coefficients. A key aspect is the use of a block-floating-point representation, where mantissa from multiple subbands share a common exponent. Fielder also describes techniques for efficiently coding the bit allocation information and exponents. Special handling is applied to low-frequency coefficients to avoid audible artifacts.

110. Specifically, Fielder's bit allocation method comprises two steps: a first, "fixed" bit allocation step and a second, "adaptive" bit allocation step. EX1006, 8:24-56, 9:27-10:2, 21:3-28, 22:19-23:5, 28:30-51. When performing these two steps, Fielder generates a "variable bit-length code word for each transform coefficient, which code-word bit length is the sum of a fixed number of bits and a variable number of bits determined by adaptive bit allocation." EX1006, 8:24-31.

111. Fielder's method is based on the following principles:

- (a) ***Component variance***: more bits are allocated to transform coefficients with the greatest level of AC power.
- (b) ***Component value***: more bits are allocated to transform coefficients which represent frequency bands with the greatest amplitude or energy.

(c) *Psychoacoustic masking*: fewer bits are allocated to signal components whose quantizing errors are masked (rendered inaudible) by other signal components. This method is unique to those applications where audible signals are intended for human perception. Masking is understood best with respect to single-tone signals rather than multiple-tone signals and complex waveforms such as music signals.

EX1006, 5:61-6:14. As seen from these principles, Fielder allocates bits in a manner that accounts for “psychoacoustic masking” effects. EX1006, 2:15-60, 5:61-6:14, 6:39-45, 8:24-56, 21:3-22:18.

112. In the first bit allocation step, a “fixed number of bits are assigned to each subband based on empirical observations regarding psychoacoustic-masking effects of a single-tone signal in the subband under consideration.” EX1006, 8:31-35, 9:46-56. Then, in the second bit allocation step, “bits are added to reduce the audible noise to acceptable levels in the problem regions.” EX1006, 9:46-56. The second step is carried out in several phases:

Phase one 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. 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. 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.

EX1006, 23:13-27.

113. By performing the adaptive allocation with these phases, “[a]daptively allocated bits are added to reduce the audible noise to acceptable levels in the problem regions.” EX1006, 9:46-56. In this manner, Fielder’s fixed and adaptive bit allocation steps assign bits to frequency bands while accounting for psychoacoustic effects. EX1006, 2:15-60, 5:61-6:14, 6:39-45, 8:24-56, 9:27-10:2, 21:3-23:30, 28:30-51.

C. A POSA would have been motivated to improve Mahieux’s bit allocation method using Fielder’s second bit allocation step.

114. A POSA would have been motivated to implement a two-step method in Mahieux by simply adding Fielder’s second bit allocation step to Mahieux’s bit allocation process.

115. Both Mahieux and Fielder are in the same technical field and disclose complementary approaches to digital audio compression. Both references share the same goal of reducing the bit-rate representation of a digital audio signal while preserving the quality of the original signal. *Compare* EX1005, 2:61-68 (Mahieux’s method “enable[s] a major reduction in the bit rate while, at the same time, preserving the quality of the starting signal to the maximum extent”) *with*

EX1006, 1:14-16 (Fielder’s method “relates in general to high-quality low bit-rate digital signal processing of audio signals, such as music signals”). To achieve that goal, both references allocate bits to certain frequencies based on psychoacoustic criteria. *Compare* EX1005, 3:1-3 (Mahieux’s “principle of operation takes account of psychoauditive criteria”), *with* EX1006, 6:39-41 (Fielder “provide[s] improved psychoacoustic-masking techniques in a transform coder processing music signals”). In other words, the references are predicated on a common technique for accomplishing a common goal.

116. Mahieux focuses on psychoacoustic modeling and adaptive bit allocation based on masking thresholds, while Fielder emphasizes efficient representation of transform coefficients using block-floating-point techniques. Combining these approaches would yield a system that benefits from both advanced psychoacoustic modeling and efficient coefficient coding.

117. More specifically, a POSA would have recognized that Mahieux’s bit allocation method, while effective for identifying perceptually important frequency bands, may not fully optimize bit distribution within those bands. Two-step bit allocation methods such as Fielder’s were well understood in the art to address this limitation by providing a more granular allocation process. *See* Section V.E. The first step in Fielder’s method, similar to Mahieux’s approach, allocates bits to perceptually significant bands. However, Fielder’s second step offers a refinement

mechanism that can more precisely distribute bits among coefficients that have already been deemed important. This two-step process allows for a more nuanced allocation that can better adapt to the specific characteristics of the audio signal being encoded.

118. Additionally, the combination of Mahieux's psychoacoustic criteria with Fielder's adaptive second step would enable the system to better handle a wider range of audio signals. Different types of audio content (e.g., speech, music, ambient sounds) can have varying spectral characteristics and perceptual requirements. By incorporating Fielder's second step, Mahieux's system would gain flexibility to fine tune bit allocation based on both psychoacoustic importance and actual signal energy distribution. This would result in a more robust coding system capable of maintaining high perceptual quality across diverse audio content while still achieving low bit rates—a key goal in audio coding technology.

119. These are exactly the benefits that Fielder itself recognizes. Fielder explains that its two-step method accurately accounts for frequency bands having the “greatest spectral energy”—i.e., those that are most perceptible to the human ear—while taking into account “coding accuracy and total bit rate” constraints. EX1006, 6:17-27, 23:6-8, 23:13-29; *see also* EX1006, 9:46-56. Fielder's second step, in particular, “reduce[s] the audible noise to acceptable levels in ... problem regions” of the signal. EX1006, 9:49-52.

120. Accordingly, using a two-step process in Mahieux's bit allocation process, as taught by Fielder, would promote additional precision and accuracy in Mahieux, thereby furthering Mahieux's goal of creating a high-quality, low bit-rate signal. EX1005, 3:43-49, 6:30-46, 7:3-8:9, 11:33-12:64; EX1006, 9:46-56, 23:13-30. And, to implement Fielder's two-step process in Mahieux, a POSA simply would have added Fielder's second "adaptive" to Mahieux's bit allocation process. In other words, Fielder's second step would be performed on the transform coefficients that are assigned bits in Mahieux's method. By allocating any excess bits not already allocated by Mahieux's method using Fielder's second step, Mahieux's method would result in an even higher quality, lower bit-rate signal.⁴

⁴ There would likely be bits left over after Mahieux's allocation step given that "bits can only be distributed as integers," whereas the calculated number of bits is "generally not" limited to integers. EX1008, 0008. In any event, even if, in some scenarios, Mahieux allocates all of its bits such that none remains for Fielder's second step, Fielder accounts for this by reserving some bits for the second step. *See* EX1006, 9:46-49 (explaining that, after the first step, the system "remov[es] one or more bits from each transform coefficient across the spectrum"). A POSA would have used this same technique with Mahieux for all the reasons explained above—i.e., to achieve a high-quality, low bit-rate signal.

121. That combination would involve nothing more than applying a known technique (Fielder's second bit allocation step) to a known method ready for improvement (Mahieux's bit allocation process) to yield predictable results. Equivalently, combining Mahieux and Fielder would have used a known technique (Fielder's second bit allocation step) to improve a similar method (Mahieux's bit allocation process) in the same way.

122. A POSA likewise would have known how to combine Mahieux and Fielder and would have done so with a reasonable expectation of success. The combination would not require any fundamental changes to the underlying principles of either method, and instead represents a straightforward integration of complementary techniques. The predictable result would be a system that maintains Mahieux's psychoacoustic-based allocation while gaining the ability to more finely distribute bits among perceptually important coefficients, as demonstrated in Fielder. This improvement aligns with the established goal in audio coding of maximizing perceived quality while minimizing bit rate, and would be expected to yield benefits across a wide range of audio signals and operating conditions.

123. A POSA would have effectuated the combination using the conventional hardware described in the references. Both Mahieux and Fielder use conventional audio-processing encoders, algorithms, and quantizers to allocate bits

for digital audio signals. EX1005, 2:13-56, 3:43-49, 6:49-58, 11:29-12:29; EX1006, 12:41-14:58. And Fielder describes a “basic hardware architecture” for audio encoding, including an analog-to-digital converter (ADC), digital-signal processor (DSP), RAM, and EEPROM. *See* EX1006, 12:41-42, 12:64-68, 13:2-19.

124. A POSA would have understood that the “quantizers” described in Mahieux and Fielder would have been implemented using such hardware and programming. EX1005, 2:13-56, 3:43-49, 6:49-58. Indeed, I note that Mahieux states that its invention can be implemented in the European DAB (Digital Audio Broadcasting) Project, a well-known and computationally complex system. EX1005, 1:23-28. Accordingly, Mahieux’s system possesses the computational resources required to perform a two-step bit allocation. The combination thus would have amounted to a simple implementation of a combined software algorithm that would have controlled the processing performed by the DSP to allocate bits. EX1005, 12:47-63, 13:2-19. Accordingly, a POSA would have had a reasonable expectation of success that the combination would continue to operate as desired.

D. Mahieux in view of Fielder renders claim 1 obvious.

- 1. Mahieux discloses “[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:” [1P].**

125. I have been informed that the preamble to a claim is generally not limiting. To the extent that the preamble language is deemed limiting, however, Mahieux discloses the preamble.

126. Mahieux describes a “bit allocation device enabling an adaptive quantization of a digital audio signal.” EX1005, 1:17-22; *see also* EX1005, Title, Abstract, 1:61-68, 3:17-32. The signal quantized by Mahieux’s system has spectral and temporal structure. Indeed, Mahieux explains that it was well known to process analog audio signals having spectral and temporal structure using a “time/frequency transformation.” EX1005, 1:46-53, 2:13-34. Mahieux’s device includes a “transformation module 11” that, consistent with the prior art, decomposes an audio signal having spectral and temporal structure by “cut[ting] up” the signal (i.e., buffering it)⁵ “into temporal blocks” (i.e., the claimed

⁵ I understand that the BPAI determined that buffering is “either inherent or obvious to one of ordinary skill in the art[.]” EX1002, 253. I agree—as Fielder makes clear, a POSA would have understood that dividing an audio signal into

“frames”) and then subjecting the signal “to a time/frequency transformation.”

EX1005, 1:18-22, 1:46-48, 5:45-46; *see also* EX1005, 2:13-34, 4:47-50, 5:20-37, 16:7-10, FIG. 1. The resulting signal is “constituted by a plurality of spectral components” that are “distributed among a plurality of adjacent critical bands.”

EX1005, 13:34-42; *see also* EX1005, Abstract, 3:17-32, 6:17-27.

127. The spectral components are ultimately fed into “bit allocation module 16,” which determines “the number of bits R available for the expression of each of the coefficients $y(k)$ of the transformed signal.” EX1005, 6:30-37; *see also* EX1005, 11:65-12:11, 12:43-56, FIGs. 1, 7.

128. Finally, the output of Mahieux’s bit allocation module is fed into “quantizer 17,” which quantizes the signal so as to reduce the bit rate “while at the same time preserving the quality of the starting signal to the maximum extent.”

temporal blocks requires buffering the signal. *See* EX1006, 13:4-10 (describing analog-to-digital converter that “receives and buffers [a] serial stream of digitized samples, groups the samples into blocks, performs the calculations required to transform the blocks into the frequency domain, encodes the transform coefficients, formats the code words into a data stream, and transmits the encoded signal through serial data path 206”); *see also* EX1006, FIG. 1a (element 102 showing a “buffer”).

EX1005, Abstract, 6:30-33; *see also* EX1005, 1:18-22, 6:49-51. Thus, Mahieux discloses all aspects of the preamble of claim 1.

2. Mahieux discloses “a step for dividing at least one frequency interval into a plurality of frequency-units;” [1A].

129. Mahieux discloses this limitation. As explained above, Mahieux’s transformation module decomposes an audio signal into spectral components by subjecting the signal “to a time/frequency transformation.” *See* Section VIII.D.1. The resulting output is a frequency “spectrum” including the totality of the spectral components for the signal (the claimed “frequency interval”). EX1005, 3:17-32, 3:43-49, 5:38-44, 6:40-46, 6:66-7:15, 9:31-59, 12:47-56, 13:1-9, 13:34-42. The spectrum is then “cut up” or “divided” “into frequency bands” (the claimed “frequency-units”). EX1005, 1:17-21, 3:17-32, 7:13-15; *see also* EX1005, 3:43-49, 5:38-44, 6:40-46, 9:31-59, 12:47-56, 13:1-9, 13:34-42.

130. These frequency bands are depicted as B₁, B₂, and B₃ in Figure 5 below:

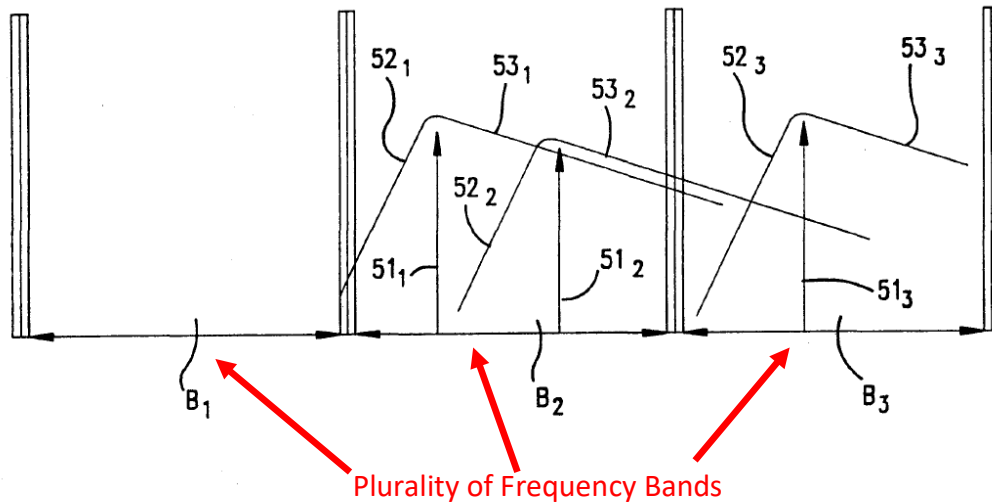


FIG.5

EX1005, FIG. 5 (annotated).

EX1005, 7:13-15. Mahieux provides examples in which the spectrum is divided into 24 and 50 frequency bands. EX1005, 7:13-15; *see also* EX1005, 9:39-42.

Accordingly, Mahieux describes dividing a frequency interval into a plurality of frequency-units as recited by the claim. EX1005, 3:17-32, 3:43-49, 6:30-58, 7:13-15, 9:39-42.

131. Even to the extent the limitation is step-plus-function, Mahieux's disclosure is identical to the corresponding acts disclosed in the '757 patent. The '757 patent describes grouping transformed spectral samples into units, just as Mahieux does. EX1001, 1:37-40, 4:21-24, FIG. 6; EX1005, 1:17-21, 3:17-32, 3:43-49, 5:38-44, 6:40-46, 7:13-15, 9:31-59, 12:47-56, 13:1-9, 13:34-42.

Accordingly, Mahieux discloses this limitation regardless of whether it is a step-plus-function limitation. At the very least, Mahieux's description of dividing a

frequency interval is equivalent to the corresponding acts disclosed in the '757 patent specification.

3. Mahieux discloses, or at least renders obvious, “a step for obtaining a representative of each frequency-unit obtained;” [1B].

132. Mahieux discloses this limitation, or at least renders it obvious. As I previously explained, Mahieux determines spectral components for the spectrum of a signal. These spectral components are grouped into frequency bands. EX1005, 3:17-32, 7:13-15. These bands are represented by “transform coefficients,” which represent the magnitudes of the frequencies within each band. EX1005, Abstract, 3:17-32, 3:43-49, 5:20-44, 6:30-58, 7:3-8:9, 10:45-49, 11:33-12:64. The “transform coefficients are defined by” the following equation:

$$y(m,k) = \sum_{n=0}^{N-1} x_m(n) \cdot h(N-1-n) \cdot \cos(2\pi(k + \frac{1}{2})(n + n_0)/N)$$

$$k = 0, \dots, N-1.$$

EX1005, 5:20-44.

In the equation above, N is the size of the transform block, h(n) is the block weighting window, and m is the block transform number. EX1005, 5:20-44.

133. Mahieux explains that “application of mathematical transformation achieves a concentration of the energy of the source signal on the biggest coefficients.” EX1005, 1:46-68. Accordingly, a POSA would have understood that

Declaration of Dr. Tim A. Williams
U.S. Patent No. 6,339,757

Mahieux's coefficients represent the energy of the frequency bands and are thus "representative" of the bands. In fact, I understand that the BPAI interpreted the claimed representative to simply mean "a characteristic representative of the signal." EX1002, 254 (cleaned up). In my view, Mahieux's transform coefficients are just that—they are characteristics representative of the signal. Additionally, Mahieux's coefficients can be used to calculate the power of each frequency band by squaring them. These power values are likewise representatives.

134. Finally, by calculating the coefficients and power values, Mahieux "obtains" them. The '757 patent similarly describes calculating representatives by calculating the logarithm of the maximum amplitude in each frequency-unit (step 2 in Figure 1) or by averaging the amplitudes in each unit. EX1001, 3:50-59, FIG. 1.

135. And, for all the same reasons, even to the extent the limitation is step-plus-function, Mahieux's calculation of its coefficients renders obvious the corresponding acts disclosed in the '757 patent. The '757 patent describes calculating a variance from the maximum spectral sample amplitude or from an average of the spectral components within the unit. EX1001, 3:50-59, FIG. 1 (numeral 2). And Mahieux calculates coefficients for all of its spectral components, which would include those having a maximum spectral sample amplitude. EX1005, Abstract, 1:46-68, 3:17-32, 3:43-59, 5:20-44, 6:30-58, 7:3-8:9, 10:45-49, 11:33-12:64. In my view, a POSA would have understood that

Mahieux's disclosure would have rendered obvious calculating a variance in the manner that the '757 patent describes.

4. Mahieux discloses, or at least renders obvious, “a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria;” [1C].

136. Mahieux discloses this limitation, or at least renders it obvious. The entire aim of Mahieux's system is “to provide a bit allocation device ... wherein the principle of operation takes account of psychoauditive criteria.” EX1005, 3:1-3; *see also* EX1005, 2:9-12 (describing using “psychoauditive criteria to achieve [a] bit rate reduction”), 1:54-68 (explaining that Mahieux's system “tak[es] account of perceptual and psychoauditive properties that are mainly linked to the spectral nature of the sound” and the “masking of inaudible spectral components”).

137. In particular, Mahieux explains that when a frequency band is “entirely masked” (e.g., rendered “inaudible” by more prominent frequencies within the signal), “it is not necessary” to allocate bits to that band. EX1005, 10:45-49. Accordingly, Mahieux ignores bands that are below a “masking threshold corresponding to the human auditory system” and thus “eliminat[es]” them from consideration—while “preserving” bands that are above the threshold. EX1005, 1:54-68, 3:17-49, 10:45-49, 11:29-38, 12:22-29, 13:13-17.

138. This process allows Mahieux to achieve its goal of “accomplish[ing] the spectral shaping of the quantization noise according to perceptual criteria” and

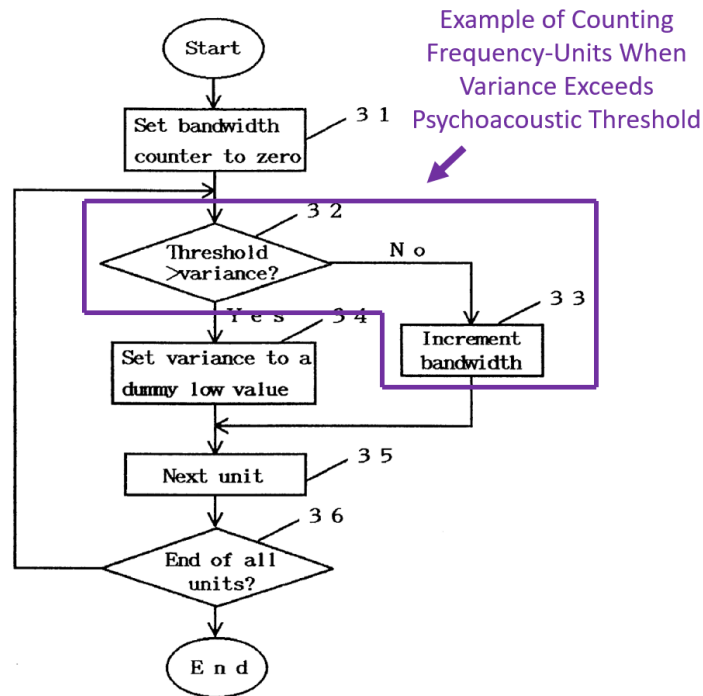
“minimizes the noise to masking threshold ration.” EX1005, 11:65-12:65. In other words, the system aims to distribute available bits among frequency bands in a way that shapes the quantization noise to align with human auditory perception. Specifically, it attempts to minimize the ratio between the quantization noise and the psychoacoustic masking threshold. The masking threshold represents the level below which sounds are inaudible due to the presence of louder sounds nearby in frequency or time. By keeping quantization noise below this threshold, the system can reduce the bit rate without introducing audible artifacts.

139. To make that determination, Mahieux compares its power values (proportional to the square of the transform coefficients) against a “masking threshold corresponding to the human auditory system.” EX1005, 8:6-9, 13:13-17. In this manner, Mahieux uses its coefficients and power values (the claimed “representatives”) to determine which bands are audible and, thus, which bands are available for bit allocation. *See* EX1005, 6:63-65 (“The function of this module is to eliminate the lines that are inaudible because of masking phenomena, so that only perceptually useful information is transmitted.”).

140. Turning to the “counting” aspect of the limitation, the ’757 patent provides an example in which a “bandwidth counter” is used to count each frequency-unit whose variance exceeds a “human hearing threshold.” EX1001, 4:29-35. If a unit exceeds the threshold, the counter “is incremented,” as shown in

step 33 of Figure 2 below:

F i g . 2



EX1001, FIG. 2 (annotated).

See also EX1001, 4:39-42. The result of this counting step, the patent explains, is to “eliminate units that are irrelevant according to psychoacoustic criteria.”

EX1001, 4:46-48.

141. Mahieux does not expressly disclose “counting” by incrementing a counter, but I understand that the Applicant acknowledged during prosecution that “there is no mention of a bandwidth counter” in the claims. See EX1002, 225-26. And a POSA would understand that Mahieux counts frequency bands that exceed a masking threshold (and thus that are available for bit allocation), as explained above. In fact, Mahieux eliminates the uncounted bands in much the same way as

described in the '757 patent.

142. To the extent that the Board determines that affirmatively counting—for example, by incrementing a counter—is required by the limitation, in my view, a POSA would have found it obvious to do so in Mahieux's system. There are only a finite number of ways to keep track of frequency bands that are perceptibly significant (sorting into perceptible and non-perceptible groups and then summing the number of bands in each group, counting the perceptible bands, or counting the non-perceptible bands), and one known way of counting in the art was to use a counter. *See, e.g.*, EX1038, 13:30-33 ("Counter 240 is a well known digital counter that counts to a maximum value, then overflows to a zero value, then counts back up to the maximum value."); EX1042, 9:11-12 ("To encode, the number of MPSs in a run are counted by a simple counter."). And so, using a counter in Mahieux's system would have at least been obvious to try. That approach would have been straightforward and would have provided an effective way to determine the frequency bands that are to be allocated bits.

143. Additionally, I am informed that Patent Owner has taken a broad view of what affirmatively counting encompasses in the parallel district court litigation. I am informed that Patent Owner mapped the "counting" step to the following excerpt of pseudocode in the ATSC A/52:2018 Digital Audio Compression Standard:

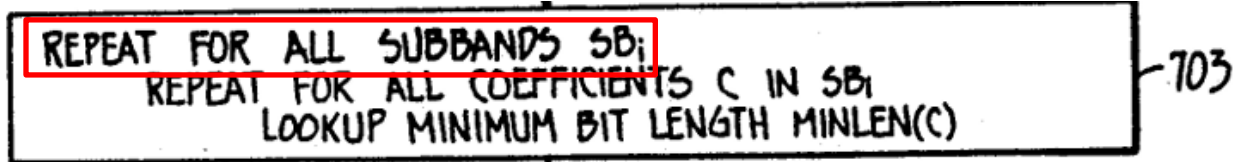
Pseudo Code

```
bndstrt = masktab[start] ;
bndend = masktab[end - 1] + 1 ;
if (bndstrt == 0) /* For flw and lfe channels */
{ /* Note: Do not call calc_lowcomp() for the last band of the lfe channel, (bin = 6) */
    lowcomp = calc_lowcomp(lowcomp, bndpsd[0], bndpsd[1], 0) ;
    excite[0] = bndpsd[0] - fgain - lowcomp ;
    lowcomp = calc_lowcomp(lowcomp, bndpsd[1], bndpsd[2], 1) ;
    excite[1] = bndpsd[1] - fgain - lowcomp ;
    begin = 7 ;
    for (bin = 2; bin < 7; bin++)
    {
        if ((bndend != 7) || (bin != 6)) /* skip for last bin of lfe channels */
        {
            lowcomp = calc_lowcomp(lowcomp, bndpsd[bin], bndpsd[bin+1], bin) ;
        }
        fastleak = bndpsd[bin] - fgain ;
        slowleak = bndpsd[bin] - sgain ;
        excite[bin] = fastleak - lowcomp ;
        if ((bndend != 7) || (bin != 6)) /* skip for last bin of lfe channel */
        {
            if (bndpsd[bin] <= bndpsd[bin+1])
            {
                begin = bin + 1 ;
            }
        }
    }
}
```

EX1037, 67 (excerpted and annotated).

144. I have reviewed the cited pseudocode and concluded that it does not perform any counting. As far as I can tell, Patent Owner is pointing to the pseudocode's iteration through the available frequency-units ("bins") in a "for" loop as an example of "counting." EX1035, 3. To the extent that the limitation encompasses such counting, that is how a POSA would have implemented Mahieux's bit allocation process. To allocate bits to Mahieux's bands, the bands must be iteratively cycled through.

145. Indeed, Fielder expressly discloses iterating through frequency sub-bands to determine transform coefficients, as shown below:



EX1006, FIG. 7 (excerpted and annotated).

146. Using a similar iterative process in Mahieux would have been the most straightforward way to implement Mahieux's system in software, as the software would need a way to account for and loop through all frequency bands that exceed Mahieux's masking threshold. The use of such a counter in Mahieux would not have changed any aspect of its system. Accordingly, in my view, utilizing a counter to actively count available frequency bands represents a mere design choice that would have yielded predictable results.

147. The above analysis applies with equal force even to the extent the limitation is step-plus-function. As explained above, it would have been obvious to implement Mahieux to count frequency-units by incrementing a counter, as disclosed in the '757 patent specification. EX1001, 4:29-48, 4:51-53, FIG. 1 (numeral 3), FIG. 2 (numerals 32-33), FIG. 3 (numeral 41); EX1005, 1:54-68, 2:9-12, 3:1-3, 3:17-49, 6:63-65, 8:6-9, 10:45-49, 11:29-38, 12:22-29, 13:13-17.

- 5. Mahieux discloses, or at least 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; and" [1D].**

148. Mahieux discloses this limitation, or at least renders it obvious. As

explained above, Mahieux counts—or at least renders obvious counting—frequency bands that are non-masked and thus that include perceptible frequencies and, therefore, that are available for bit allocation. *See* Section VIII.D.4. Mahieux then allocates bits to those bands using an approximate mathematical model.

149. An example of such a model “for allocating bits to each band” is shown in module 72 of Figure 7 below:

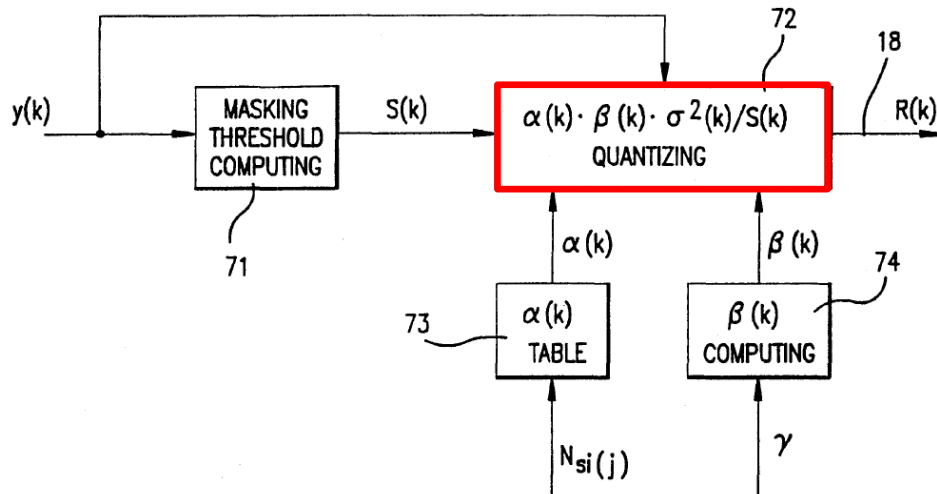


FIG.7

EX1005, FIG. 7 (annotated).

EX1005, 5:10-11. As shown above, the model is based on the following ratio:

$\alpha(k) \cdot \beta(k) \cdot \sigma^2(k)/S(k)$. EX1005, 12:3-11.

150. In this equation, “ $\sigma^2(k)$ 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.” EX1005, 12:12-14. The function $\alpha(k)$ corresponds to

a performance factor based on table 73 as seen from Figure 7. “This table is addressed by the number of non-zero coefficients.” EX1005, 12:25-26. The function $\beta(k)$ “force[s] the allocation of the bits to grant a greater number of bits to the zones of the spectrum that contain peaks.” EX1005, 12:49-51. Specifically, $\beta(k)$ “takes account of the spectral spread γ ” and “indicates whether the spectrum is concentrated or not.” EX1005, 12:30-42. “For the allocation of the bits, γ 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.” EX1005, 12:43-46. Mahieux provides the following equation for calculating γ :

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

EX1005, 12:35-42.

151. Mahieux describes allocating bits based on this mathematical model in the following manner:

In each of these four frequency bands, the function $\beta(k)$ is computed according to the value of γ 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.

EX1005, 12:47-56.

152. Further, as I previously discussed with reference to limitation [1C], Mahieux counts the number of frequency bands that will be assigned bits after spectral components have been eliminated based on a psychoauditive criterion. *See* Section VIII.D.4. For example, a “spectrum is divided into 24 critical bands, B_1 , B_2 , B_3 and multiplied by the ear transmission factor a_o for each of said bands.” EX1005, 7:13-15. “If a frequency band is entirely masked,” however, “it is not necessary to transmit a code word for the band.” EX1005, 10:45-49. That is, the band will not receive any allocated bits because the spectral components in the band are inaudible. EX1005, 10:45-49; *see also* EX1005, 4:35-42, 5:66-6:6, 6:63-65, 12:57-63. The band will not be counted for the purposes of assigning bits. EX1005, 10:45-49; *see also* EX1005, 4:35-42, 5:66-6:6, 6:63-65, 12:57-63.

153. As explained above, Mahieux then counts the remaining bands in the spectrum and uses those bands for bit allocation. Specifically, the remaining bands with “non-zero coefficients” are assigned bits based on the “number of coefficients to be quantized $N_{st}(j)$ ” where “ j is the number of the band to which $y(k)$ [the coefficient] belongs.” EX1005, 11:29-38. “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.” EX1005, 11:44-62, FIG. 6. In other words, Mahieux uses the counted number of frequency bands to allocate bits. EX1005, 10:45-49; *see also* EX1005, 4:35-42, 5:66-6:6, 6:63-65, 12:57-63. It therefore would have been obvious to a POSA that Mahieux’s mathematical model only accounts for the frequency bands that have been counted.

154. Thus, a POSA would have understood that Mahieux allocates bits for each available frequency band by using an approximate mathematical model using the counted number of available bands.

155. Even to the extent the limitation is step-plus-function, Mahieux’s bit allocation using a mathematical model is equivalent to the corresponding acts disclosed in the ’757 patent specification. EX1001, 4:49-5:11, FIG. 1 (numeral 4), FIG. 3 (element 43); EX1005, 10:45-49, 12:3-11, 12:25-26, 12:47-56. In both Mahieux and the ’757 patent, the mathematical model allocates bits to the most energetic frequency bands and based on masking criteria. Thus, a POSA would have found the two models to be equivalent.

6. Mahieux in view of Fielder teaches “a second bit allocation step for determining further bit allocation for each of said available frequency-units based on psychoacoustic criteria.” [1E].

156. As explained above, Mahieux describes a single-step bit allocation process using a mathematical model. As also explained above, a POSA would have

been motivated to use Fielder's two-step method to achieve greater granularity in Mahieux's system. *See* Section VIII.C. In addition, the need for two-step bit allocation was well recognized by a POSA as a method for dealing with the universal issue that "bits can only be distributed as integers." EX1008, 0007. And, to implement Fielder's method in Mahieux, a POSA would have added Fielder's second bit allocation step.

157. Fielder's second step is itself performed in an iterative fashion and has "four phases":

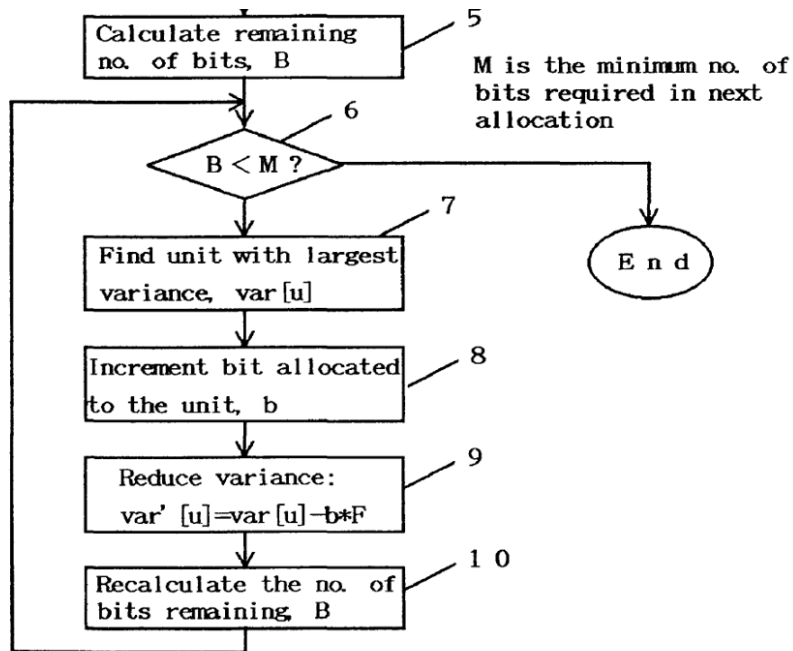
Phase zero is an initialization process for the remaining phases. Phase one assigns bits, up to a maximum of four per transform coefficient, to the coefficients within the same critical band of those frequency components with the greatest spectral energy. If all allocable bits are assigned during phase one, the allocation process stops. 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. 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.

EX1006, 23:13-30. Fielder explains that this method accounts for "psychoacoustic masking" such that "fewer bits are allocated to signal components whose quantizing errors are masked (rendered inaudible) by other signal components."

EX1006, 6:7-10.

158. Indeed, a POSA would have understood that the frequency components having “the greatest spectral energy” described in Fielder’s phase one above are the audible components that are most perceptible to the human ear. *See* EX1006, 5:61-6:14 (explaining that “more bits are allocated to transform coefficients which represent frequency bands with the greatest amplitude or energy”), 8:45-51, 23:13-29. That, after all, is the point of Fielder’s adaptive bit allocation step—i.e., to allocate bits only to the most audible frequencies in order to reduce quantization error for those frequencies. *See* EX1006, 9:46-56 (explaining that bits are “added to reduce the audible noise to acceptable levels in [any] problem regions”), 5:61-6:14 (explaining that “fewer bits are allocated to signal components whose quantizing errors are masked (rendered inaudible) by other signal components”). Accordingly, Fielder’s adaptive bit allocation method is based on “psychoacoustic criteria,” as required by the limitation.

159. I note that Fielder’s iterative process mirrors the ’757 patent’s method:



EX1001, FIG. 1 (excerpted).

In this example, the '757 patent considers a remaining number of bits to allocate at step 5. EX1001, 3:62-63. In steps 7 and 8, the '757 patent describes assigning additional bits to the unit with the largest variance. EX1001, 4:4-10. The '757 patent describes examples of variance as being “the maximum spectral sample amplitude in the unit” or “an average of the spectral components within the unit.” EX1001, 3:50-55. Fielder similarly assigns bits to the critical bands “with the greatest spectral energy.” EX1006, 23:13-29; *see also* EX1006, 8:45-51. Fielder therefore describes a second bit allocation that has the same band energy consideration as the '757 patent. EX1006, 23:13-29; *see also* EX1006, 8:45-51.

160. Even to the extent the limitation is step-plus-function, Fielder's adaptive allocation step is equivalent to the corresponding acts disclosed in the

'757 patent specification. EX1001, 3:63-4:18, FIG. 1 (numerals 6-10); EX1006, 5:61-6:14, 9:46-56, 23:13-30. As I demonstrated above, both Mahieux and the '757 patent iteratively allocate bits based on psychoacoustic criteria. Thus, in my view, the two are equivalent.

161. Finally, as discussed above, a POSA would have been motivated to employ Fielder's second bit allocation step in Mahieux's method, as doing so would achieve greater quantization precision and accuracy. *See* Section VIII.C. In that combination, Fielder's method would be used to allocate bits to the available frequency bands that Mahieux counts, as explained above. *See* Sections VIII.D.4-5.

E. Mahieux in view of Fielder renders claim 2 obvious.

162. The combination of Mahieux and Fielder renders obvious a "[m]ethod 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," as recited in claim 2.

163. As explained above with respect to limitation [1D], Mahieux discloses allocating bits to frequency bands using an approximate mathematical model. *See* Section VIII.D.5. As also explained above, the allocation process accounts for the "masking of inaudible spectral components" and "eliminate[s]" from consideration the "masked lines" corresponding to "inaudible frequencies." EX1005, 1:54-68, 4:67-68, 6:1-5; Sections VIII.D.4-5.

164. Mahieux also accounts for masked lines from within a single band (i.e., “in-band”). In particular, Mahieux describes “intra-band masking”:

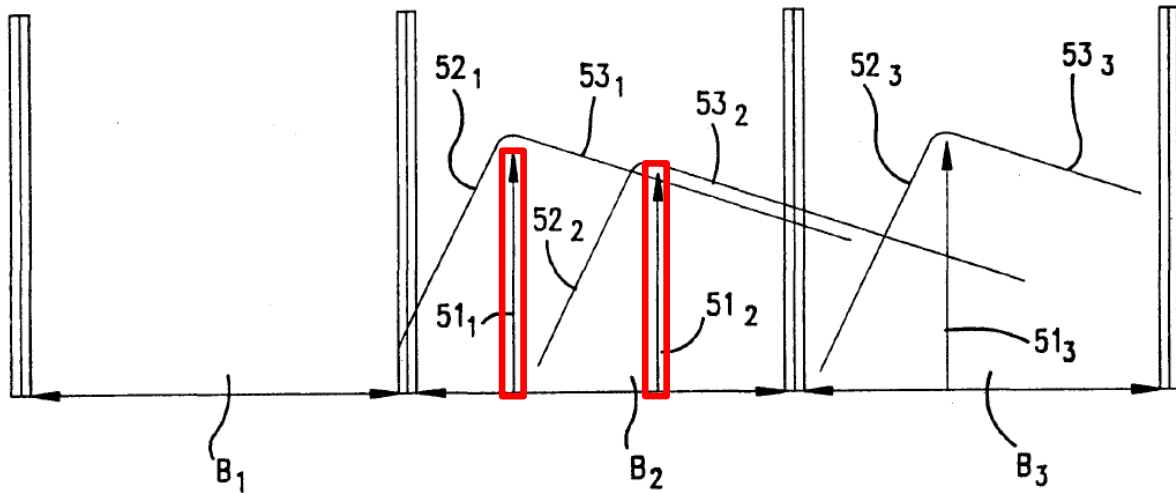


FIG.5

EX1005, FIG. 5 (annotated).

EX1005, 7:16-32. Mahieux explains that “intra-band masking” “corresponds to the total masking effect of all the lines 51₁, 51₂ within one and the same band B₂.”

EX1005, 7:19-23. And, as explained above, “[i]f a frequency band is entirely masked,” it is “not necessary” to allocate bits for the band. EX1005, 10:45-49.

165. Mahieux’s mathematical model accounts for masking and is therefore “derived from” an in-band masking effect, as required by the claim. Indeed, the “masking threshold” is captured by $S(k)$, which appears in Mahieux’s mathematical model: $\alpha(k) \cdot \beta(k) \cdot \sigma^2(k) / S(k)$. EX1005, 12:3-11. Additionally, $\alpha(k)$,

which also appears in the model, is a function of $N_{st}(j)$, which is “the number of non-masked coefficients in the band j .” EX1005, 9:58-59. Mahieux’s mathematical model thus accounts for in-band masking effects.

166. And, because the model accounts for in-band masking effects, it leads to an error spectrum that follows more closely to the original signal spectrum, as required by the claim. In particular, Mahieux explains that its model and bit allocation method are “designed to accomplish the spectral shaping of the quantization noise according to perceptual criteria” by “minimiz[ing] the noise to masking threshold ratio.” EX1005, 11:65-68; *see also* EX1005, 6:38-39. Mahieux’s method thus codes “pure sounds contained in the signal ... with higher precision” such that the masking threshold “is lower than in the case of a noise spectrum.” EX1005, 12:51-56. This method results in a signal having coefficients that are “very close to the true spectrum.” EX1005, 5:38-44, 12:57-63; *see also* EX1005, 2:61-68 (explaining that Mahieux’s system “preserv[es] the quality of the starting signal to the maximum extent”).

167. In other words, by accounting for in-band masking effects, Mahieux minimizes noise for frequencies that are most perceptible, thereby yielding an error spectrum that follows more closely to the original signal spectrum for those frequencies. Lowering the error spectrum in this fashion mirrors the ’757 patent’s disclosure. *See* EX1001, 5:24-27 (“[I]n the higher frequency end, the error

spectrum ... is lower and this accounts for the improvement in the noise control at high frequency.”).

168. Thus, Mahieux in view of Fielder renders claim 2 obvious.

F. Mahieux in view of Fielder renders claims 3, 7, 11, and 14 obvious.

169. The combination of Mahieux and Fielder renders obvious a “[m]ethod according to claim 2, wherein each frequency-unit is represented by a single spectral component,” as recited in claim 3. The combination also renders obvious claims 7, 11, and 14, which similarly recite each frequency-unit being represented by a single spectral component. *See* EX1001, Claims 7, 11, 14. Claim 7 depends from claim 6, which depends from claim 1. Claim 11 depends from claim 10, which also depends from claim 1. And claim 14 depends directly from claim 1.

170. As explained above, Mahieux describes dividing a signal’s frequency spectrum into various bands and obtaining transform coefficients and power values (the claimed “representatives”) for each band. *See* Sections VIII.D.2-3. Further, Mahieux depicts an example in which one of those bands (B₃) includes a single spectral component (51₃):

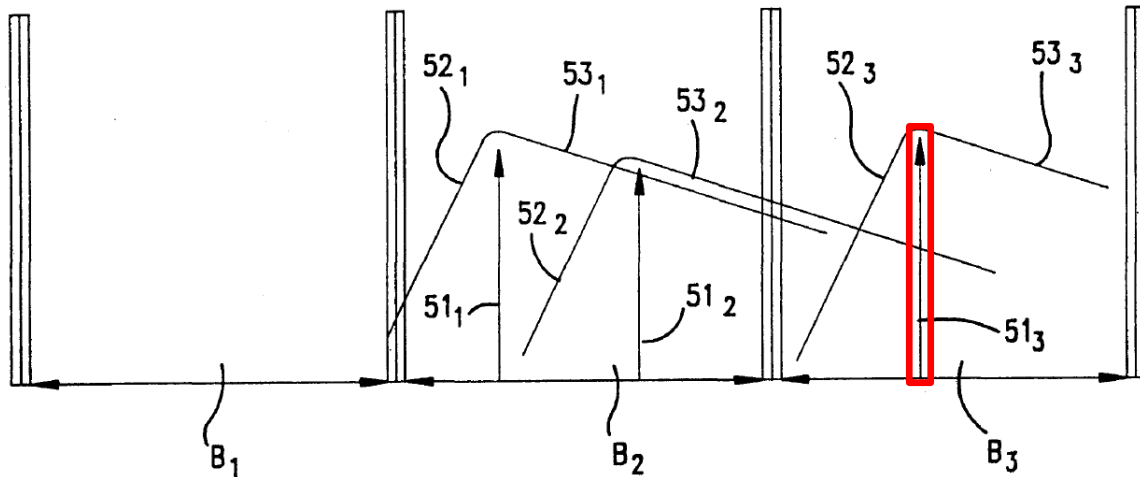


FIG.5

EX1005, FIG. 5 (annotated).

The unit's representative is the transform coefficient or power value for the single spectral component 51₃.

171. Mahieux explains that a spectrum can be divided into several bands of “equal width.” EX1005, 9:13-17, 12:43-46. Given that Mahieux expressly discloses at least one band having a width that accommodates a single spectral component, and given that Mahieux explains that all bands can have the same width, a POSA would have understood that all bands can include just a single spectral component, or at least that they can be represented by a single component. EX1005, 7:13-15, FIG. 5.

172. Thus, Mahieux in view of Fielder renders these claims obvious.

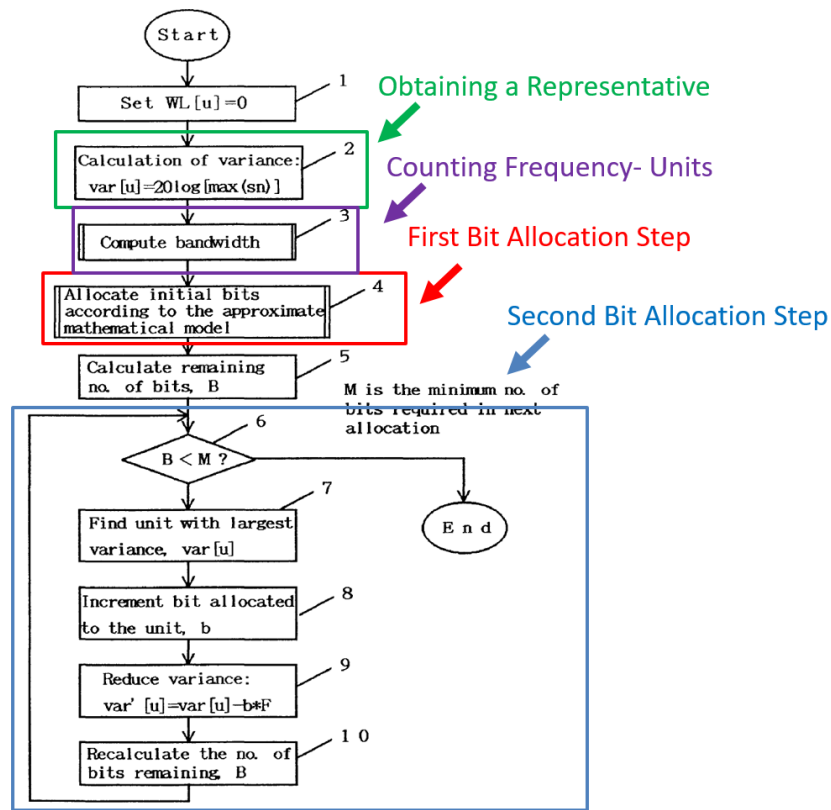
G. Mahieux in view of Fielder renders claim 6 obvious.

173. The combination of Mahieux and Fielder renders obvious 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,” as recited in claim 6.

174. I note that the '757 patent does not meaningfully discuss a “psychoacoustically weighted inverse relationship” with the representative of each frequency-unit. Rather, the Abstract states that the claimed method “iteratively increas[es] the quantization of each of the frequency intervals through a psychoacoustically weighted inverse relationship with the [sic] variance of each of the frequency intervals.” EX1001, Abstract.

175. The only disclosure in the '757 patent bearing on the second bit allocation step is the description of steps 6-10 in Figure 1:

F i g . 1



EX1001, FIG. 1 (annotated).

EX1001, 3:62-4:18. As steps 7-8 show, frequency-units having the “largest variance” are allocated the most bits. Thus, as the variance of a frequency-unit increases, the quantization for the unit increases and the corresponding error decreases. In other words, variance and quantization error are inversely related, as Fielder itself explains. EX1006, 1:33-39 (explaining that quantization error is “inversely proportional to the number of bits available to represent the signal sample quantization”).

176. Accordingly, adverse psychoacoustic effects are reduced through the

iterative bit allocation process shown in Figure 1. The bit allocation is determined based on a psychoacoustically weighted inverse relationship with the variance of each frequency-unit.

177. As explained above, that is what Fielder discloses. *See* Section VIII.D.6. Fielder's second bit allocation step considers "psychoacoustic masking" so that "fewer bits are allocated to signal components whose quantizing errors are masked (rendered inaudible) by other signal components," and "more bits are allocated to transform coefficients" and power values (i.e., the claimed "representatives") having "the greatest amplitude or energy." EX1006, 5:61-6:14. This method, Fielder explains, "reduce[s] the audible noise to acceptable levels in the problem regions." EX1006, 9:46-56. And, like the '757 patent, Fielder iteratively assigns bits in this manner "until all allocable bits are assigned." EX1006, 25:1-13. Fielder's adaptive bit allocation performs the same band identification and iterative bit assignment disclosed in the '757 patent.

178. Thus, Mahieux in view of Fielder renders claim 6 obvious.

H. Mahieux in view of Fielder renders claim 10 obvious.

179. The combination of Mahieux and Fielder renders obvious 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," as recited in claim 10.

180. As explained above with respect to limitations [1B] and [1C], Mahieux obtains transform coefficients and power values to determine whether a particular frequency band includes spectral components that meet psychoacoustic criteria. *See* Sections VIII.D.3-4.

181. In particular, if a frequency band is “inaudible” or “entirely masked,” “it is not necessary” to allocate bits to the band. EX1005, 10:45-49; *see also* EX1005, 4:35-42, 5:66-6:6, 6:63-65, 12:57-63. Mahieux therefore compares its power values (proportional to the square of the transform coefficients) (the claimed “representatives”) against a “masking threshold corresponding to the human auditory system” (the claimed “human hearing threshold”)—i.e., it ignores bands that are “below [the] threshold” and “preserv[es]” bands that are above the threshold. EX1005, 8:6-9, 13:13-17. The preserved bands are “representative of the sensibility of the human ear.” EX1005, 13:1-9.

182. By accounting for bands that are above the masking threshold, Mahieux counts—or at least renders obvious counting—bands that exceed a human hearing threshold, as required by the claim. Once again, Mahieux’s disclosure is very similar to the ’757 patent’s. *See* EX1001, 4:46-48 (explaining that using a “hearing threshold will eliminate units that are irrelevant according to psychoacoustic criteria”).

183. Thus, Mahieux in view of Fielder renders claim 10 obvious.

I. Mahieux in view of Fielder renders claim 16 obvious.

184. The combination of Mahieux and Fielder renders obvious a “[m]ethod according to claim 1, wherein each frequency-unit is obtained by dividing said spectral and temporal structure with at least one frequency interval,” as recited in claim 16.

185. As explained above with respect to limitations [1P] and [1A], Mahieux discloses bit allocation with respect to an audio signal having spectral and temporal structure. *See* Sections VIII.D.1-2. Mahieux divides the signal into “blocks of samples” where a “time/frequency transformation and a predictive and adaptive coding of the most significant coefficients of each block” occur. EX1005, 2:13-23. Upon transforming the signal, Mahieux determines a frequency “spectrum,” which is then divided to identify “frequency bands.” EX1005, 3:17-32, 3:43-49, 6:30-58, 7:13-15, 9:39-42. Dividing of the “totality of the spectrum” into “a set of adjacent bands” corresponds to obtaining of a frequency-unit using at least one frequency interval. EX1005, 3:17-32.

186. Mahieux further determines spectral components in the frequency domain which are represented by coefficient and power values. EX1005, 3:17-32, 5:63-65; *see also* EX1005, 1:46-68, 2:13-34, 3:36-40, 5:20-36, 6:40-46, 12:57-63; Section VIII.D.3. Mahieux allocates a “specific number of bits for the expression of the coefficients of each frequency band of a transformed digital audio signal.”

EX1005, Abstract. The “allocation consists notably in the assigning, to each band in a set of adjacent bands covering the totality of the spectrum of the transformed signal, of a number of specific bits for the expression of the transformed coefficients of said signal.” EX1005, 3:17-32. In this manner, Mahieux discloses obtaining a frequency-unit (i.e., Mahieux’s “frequency band”) by dividing a signal’s spectral and temporal structure with at least one frequency interval.

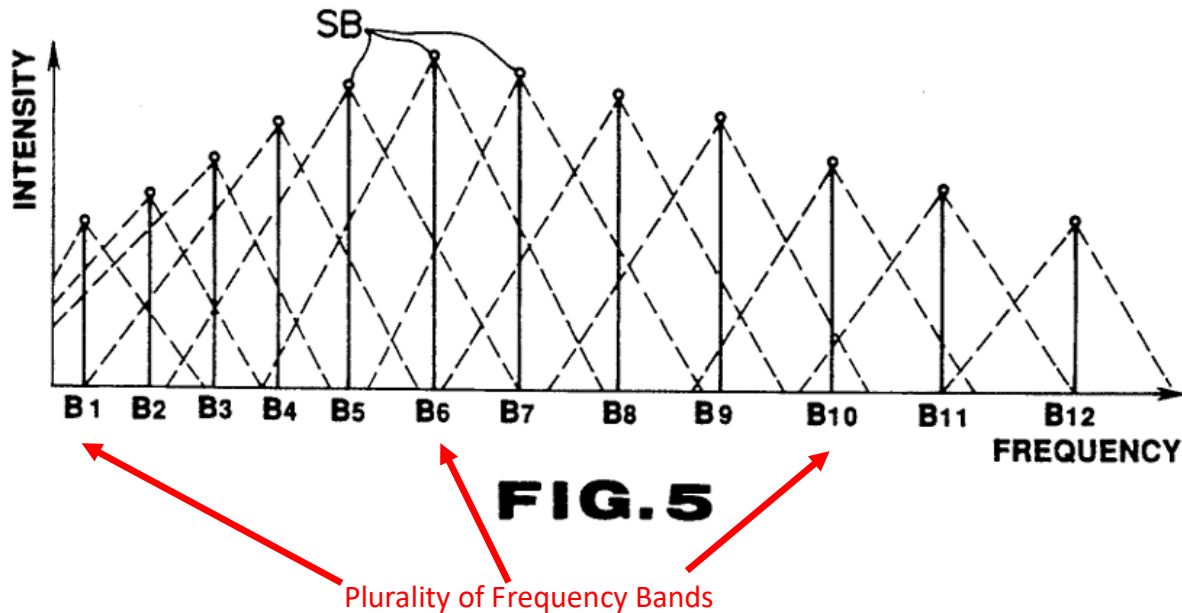
187. Thus, Mahieux in view of Fielder renders claim 16 obvious.

IX. GROUND 2: CLAIMS 1-17 OF THE ’757 PATENT IS OBVIOUS OVER THE COMBINATION OF MAHIEUX, FIELDER, AND AKAGIRI.

188. As I explain below, it is my opinion that claims 1-17 of the ’757 patent are also obvious over the combination of Mahieux, Fielder, and Akagiri.

A. Overview of Akagiri

189. U.S. Patent No. 5,241,603 (“Akagiri”) describes “an apparatus for encoding an input digital signal.” EX1007, 1:6-7, 2:21-26. Akagiri’s device transforms an audio signal into a frequency-domain representation and divides the signal “into a number of frequency bands”:



EX1007, FIG. 5 (annotated).

EX1007, Abstract, 2:32-50, 4:14-67.

190. Akagiri then determines “the energies or spectral intensities of each band” by summing either the “peak value” or “mean value” of spectral components within the band. EX1007, 4:58-68. Ultimately, bits are allocated to bands having the largest energies, thus accommodating for “masking characteristics of the human auditory sense.” EX1007, 1:32-35, 4:42-57. In this manner, Akagiri takes into account “human auditory characteristics.” EX1007, 4:46-48.

B. Motivation to combine Akagiri with Mahieux and Fielder

191. A POSA would have been motivated to use Akagiri in the Mahieux-Fielder combination to calculate Mahieux’s transform coefficients and power values (the claimed “representatives”) from spectral components’ peak values (claims 4, 8, 12, and 15) and average values (claims 5, 9, 13, and 17) within each

of Mahieux's frequency bands.

192. Both Mahieux and Fielder already obtain representatives for frequency bands and use them to allocate bits. *See* Sections VIII.D.3-6. Mahieux and Fielder consider the “biggest coefficients” and “greatest amplitude or energy” in a frequency band to allocate bits. EX1005, 1:54-60; EX1006, 6:4-6. A POSA, when assigning bits considering the “greatest amplitude or energy” in a frequency band as taught by Mahieux and Fielder, would have implemented the method using Akagiri's “peak values.” Such values indisputably represent the “greatest amplitude or energy” values. Thus, calculating representatives by determining peak amplitude values for each band is nothing more than a simple substitution of one known element (using coefficients and power values) for another (using peak values) to obtain predictable results.

193. Additionally, a POSA would have recognized that another way to represent a frequency band is to consider an average or mean value of the amplitudes within the band. EX1007, 4:58-68. For example, Honda teaches that “[a]n average amplitude ... of the signal of each band is detected for each temporal sub-interval, and a quantization bit number and a quantization step size are adaptively determined from the average amplitude for each sub-interval[.]” EX1017, Abstract. This technique not only “permit[s] reduction of the quantity of hardware used,” EX1017, Abstract, it also captures a broader spectrum of

frequencies in each band.

194. Indeed, Akagiri's approach presents a number of advantages. First, Akagiri offers flexibility by providing multiple options for representing band energy (sum of amplitude values, peak values, or mean values). This flexibility allows for adapting the calculation method to different types of audio signals or computational constraints, which would enhance the bit allocation system of Mahieux-Fielder.

195. Second, Akagiri's use of critical bands aligns well with psychoacoustic principles, as critical bands correspond to the frequency resolution of the human auditory system. This would improve the perceptual quality of the bit allocation in the Mahieux-Fielder system.

196. Third, depending on the specific method chosen (sum, peak, mean, or energy sum), Akagiri's approach would offer computational advantages in calculating the variance, which could be beneficial in real-time encoding applications.

197. Fourth, Akagiri's spectrum provides a representation of the overall spectral shape of the audio signal. This would be valuable for bit allocation decisions in Mahieux-Fielder's system, leading to improved allocation based on the signal's spectral characteristics.

198. Finally, Akagiri's approach would achieve a more accurate bit

allocation in Mahieux-Fielder that would align more closely to the original frequency spectrum in certain circumstances. For example, if the frequency band is wide and includes many spectral components, or if only a few frequency bands are used to represent the signal, a variance based on an average or mean value would more accurately reflect the frequency characteristics of each frequency band. This is because the average value would convey information corresponding to all of the spectral components within the band.

199. The above benefits provide an affirmative rationale as to why a POSA would seek to use Akagiri's teachings in the Mahieux-Fielder combination. But, at the very least, using the average or mean value of the amplitudes within a band would be obvious to try given the prevalence of that technique in the art and given that there are only a finite number of ways to represent a frequency band (e.g., peak value, mean value, median, etc.). *See* EX1016, 15:37-16:2 ("This allocation process may be based on peak-to-peak amplitude, root-mean square (RMS) amplitude, or other functions of signal level."); EX1017, Abstract ("a quantization bit number and a quantization step size are adaptively determined from the average amplitude for each sub-interval.").

200. Finally, a POSA would have known how to combine Akagiri with Mahieux and Fielder and would have done so with a reasonable expectation of success. Specifically, a POSA would have implemented Akagiri's teachings using

the DSP and programming described in Fielder. *See* Section VIII.C. This combination would have led to predictable results and would have worked as intended because Mahieux, Fielder, and Akagiri use conventional audio-processing encoders and algorithms for encoding digital audio signals. *See* Section VIII.C; EX1007, 3:44-4:41. Thus, a POSA would have had an expectation of success that the combination would continue to operate as desired.

C. Mahieux in view of Fielder and Akagiri renders claims 4, 8, 12, and 15 obvious.

201. The combination of Mahieux, Fielder, and Akagiri renders obvious a “[m]ethod 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,” as recited in claim 4. The combination also renders obvious claims 8, 12, and 15, which similarly recite the representative being a variance calculated from an amplitude of a maximum spectral component of each frequency-unit. *See* EX1001, Claims 8, 12, 15. Claim 8 depends from claim 6, which depends from claim 1. Claim 12 depends from claim 10, which likewise depends from claim 1. Claim 15 depends directly from claim 1.

202. The ’757 patent does not explain what a variance is. Of course, the term “variance” generally has a well-defined statistical meaning (e.g., the spread between numbers in a data set), but the patent appears to adopt a broader meaning. It states that the variance can be “calculated from the maximum spectral sample

amplitude in [each] unit,” and the only example provided (in step 2 of Figure 1) shows that the variance is calculated by the following equation:

$\text{var}[u]=20\log[\max(\text{sn})]$. EX1001, 3:50-52, FIG. 1. I note that the Applicant stated during prosecution that the term “variance” is used “to define a change [in] each frequency unit,” without specifying how the change is determined. EX1002, 179.

203. In view of the '757 patent's broad disclosure and the Applicant's broad statements, Mahieux, Fielder, and Akagiri render the limitation obvious. As explained above, Mahieux discloses representatives in the form of transform coefficients and power values representing amplitudes of spectral components in each frequency band. *See* Section VIII.D.3. Akagiri discloses using a peak amplitude value to represent a frequency band. In particular, Akagiri determines the “energies or spectral intensities of each band” by considering the “amplitude values A_m for the critical bands.” EX1007, 4:58-68. In one embodiment, Akagiri specifically sums the “peak value[s]” of the amplitudes in each band. EX1007, 4:58-68. The peak value is the claimed “amplitude of a maximum spectral component of each frequency-unit.” And, by summing the peak values so as to obtain “energies or spectral intensities,” Akagiri calculates the variance “from” the amplitude of a maximum spectral component, as required by the claims.

204. Finally, as discussed above, a POSA would have been motivated to employ Akagiri's method of determining peak amplitude values for each band in

the combination of Mahieux and Fielder, as doing so would constitute a simple substitution of one known element (using coefficients and power values) for another (using peak values) to obtain predictable results. *See* Section IX.B.

D. Mahieux in view of Fielder and Akagiri renders claims 5, 9, 13, and 17 obvious.

205. The combination of Mahieux, Fielder, and Akagiri renders obvious a “[m]ethod according to claim 2, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit,” as recited in claim 5. The combination also renders obvious claims 9, 13, and 17, which similarly recite the representative being a variance calculated from an average of a spectral component of each frequency-unit. *See* EX1001, Claims 9, 13, 17. Claim 9 depends from claim 6, which depends from claim 1. Claim 13 depends from claim 10, which likewise depends from claim 1. Claim 17 depends directly from claim 1.

206. Once again, the ’757 patent does not explain precisely how the variance is determined except to say that it “can be calculated from an average of the spectral components within [each] unit.” EX1001, 3:53-55. And, as explained above, the Applicant stated during prosecution that the term “variance” is used “to define a change [in] each frequency unit,” without specifying how the change is determined. EX1002, 179.

207. With that understanding in mind, Mahieux, Fielder, and Akagiri

render the claims obvious. In particular, Akagiri states that “the mean value of the amplitude values” may be summed to obtain “the energies or spectral intensities of each band,” which are ultimately used to allocate bits. EX1007, 4:58-68. The mean value of the band amplitudes is the claimed “average of a spectral component of each frequency-unit.” And, by summing the mean values so as to obtain “energies or spectral intensities,” Akagiri calculates the variance “from” the average of a spectral component of each frequency-unit, as required by the claims.

208. Finally, as explained above, a POSA would have been motivated to employ Akagiri’s method of determining an average value of the spectral components for each band in the Mahieux-Fielder combination, as doing so would capture a wider spectrum of frequencies in each of the bands. *See* Section IX.B.

E. Mahieux in view of Fielder and Akagiri renders claims 1-3, 6-7, 10-11, 14, and 16 obvious.

209. Given that Akagiri discloses the representative recited in claims 4 (“a variance which is calculated from an amplitude of a maximum spectral component”) and 5 (“a variance which is calculated from an average of a spectral component”), it also discloses the representative recited in independent claim 1. At the very least, Akagiri discloses determining the sum of peak and mean amplitude values within each band, which would qualify as a “representative”—i.e., “a characteristic representative of the signal.” EX1002, 254 (cleaned up).

210. Thus, even if the Board were to conclude that the combination of

Mahieux and Fielder does not render claim 1 obvious, the combination of Mahieux, Fielder, and Akagiri does. Accordingly, Mahieux, Fielder, and Akagiri render claims 1-3, 6-7, 10-11, 14, and 16 obvious as well.

X. GROUND 3: CLAIMS 1-3, 6-7, 10-11, 14, AND 16 OF THE '757 PATENT ARE OBVIOUS OVER THE COMBINATION OF VELDHUIS AND GALAND.

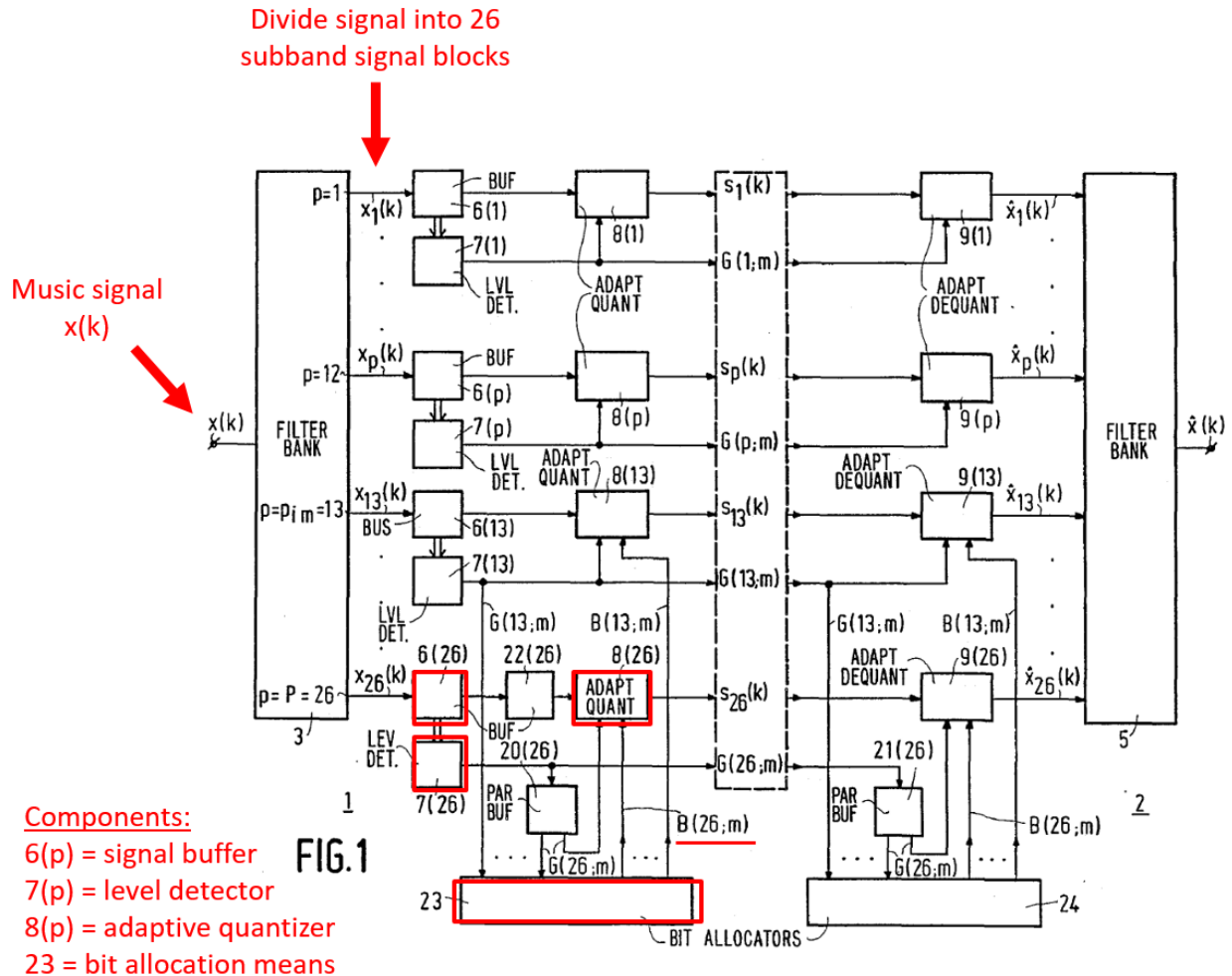
211. As I explain below, it is my opinion that claims 1-3, 6-7, 10-11, 14, and 16 of the '757 patent are obvious over the combination of Veldhuis and Galand.

A. Overview of Veldhuis

212. U.S. Patent No. 4,896,362 ("Veldhuis") discloses a system for subband coding of a digital audio signal. EX1008, Title, Abstract, 1:6-40. Veldhuis identifies shortcomings of conventional technology as follows: "[W]hile the quality of the replica of the music signal does not perceptibly differ from that of the original music signal in virtually all passages of nearly all sorts of music signals. However, in certain passages of some sorts of music signals the quantizing noise is still audible." EX1008, 1:62-2:7. Veldhuis, thus, discloses a system that reduces quantization error "without increasing the average number of bits per sample of the quantized output signals of the coder." EX1008, 2:14-20. Like the '757 patent, Veldhuis accomplishes this reduction in noise by reallocating bits based on "characteristic parameters representing the signal energy" in order to

“exploit the psychoacoustic effect of temporal masking.” EX1008, 3:49-4:52.

213. Veldhuis’s method is shown in the context of Figure 1:



EX1008, FIG. 1 (annotated).

214. First, an input “signal $x(k)$ ” having a frequency band between 0 and 22.05 kHz is applied to filter bank 3 and divided into twenty-six subbands having “band numbers $p(1 \leq p \leq P=26)$ ”:

p	f _{co} [H z]		W(p) [H z]	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

EX1008, FIG. 3.

EX1008, 5:53-59. For each subband $p(13 \leq p \leq 26)$ —i.e., the subbands having frequencies between 1,723 Hz and 22.05 kHz—a signal $x_p(k)$ is generated, and the signals “are quantized blockwise” using “known block-adaptive PCM methods” via signal buffer 6(p), level detector 7(p), and adaptive quantizer 8(p). EX1008, 5:64-68, 6:10-21. Meanwhile, subbands having band numbers $p(1 \leq p \leq 12)$ receive “a fixed number of B(p) bits per signal sample.” EX1008, 10:60-66. However, these bands are not relevant to the patentability challenge here, which, as I explain in more detail below, relies only the subbands having band numbers $p(13 \leq p \leq 26)$.

I understand that the BPAI similarly noted during prosecution that it was “proper to focus on” subbands ($13 \leq p \leq 26$) in Veldhuis’463. EX1002, 255.

215. Second, for each block “(p;m),” “a characteristic parameter $G(p;m)$ ” is determined, which represents “the signal level in th[e] block.” EX1008, 6:14-18. In block (p;m), “p” represents the subband (here, $p=13-26$) and “m” represents the block. EX1008, 5:53-68. Veldhuis describes that it “is well known,” that “the signal level can be represented by the average value of the amplitude or the power of the signal samples of a block, but also by the peak value of the amplitude of the signale [sic] samples in a block.” EX1008, 6:24-28. I agree that this was well known in the art at the time of the ’757 patent.

216. Third, the characteristic parameter $G(p;m)$ is compared to a threshold $T(p)$ “of the human auditory system for just perceiving single tones in the respective frequency ranges” so as to generate a binary comparator signal $C(p;m)$. EX1008, 11:23-36. If $G(p;m)$ is greater than or equal to the threshold, $C(p;m)$ is set to 1; otherwise $C(p;m)$ is set to 0. EX1008, 11:23-28. This comparison allows the blocks to be categorized into two groups: (a) blocks that can be heard (those “containing perceptually significant signal energy” indicated by a $C(p;m)$ value of 1); and (b) blocks that cannot be heard (those “containing no perceptually significant signal energy” indicated by a $C(p;m)$ value of 0). EX1008, 11:37-49. For the blocks that cannot be heard, a “sum S of the saved numbers of quantizing

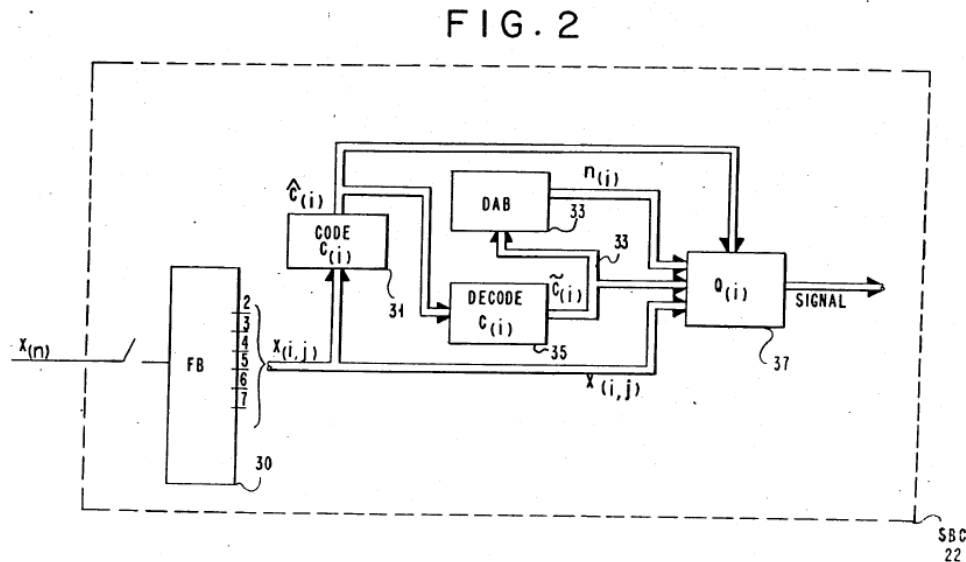
bits” is determined by “increment[ing]” the number of bits that would have otherwise been assigned to the blocks. EX1008, 11:42-49, 12:11-24, 13:51-14:23, FIG. 8.

217. Fourth, Veldhuis assigns bits to the audible subbands having $C(p;m)=1$ according to a “a predetermined allocation pattern $\{B(p)\}$,” which can comprise a particular mathematical model. EX1008, 11:44-11:58; *see also* EX1008, 13:51-14:23, FIG. 8. Veldhuis’s memory 29 stores this “predetermined allocation pattern $\{B(p)\}$ of numbers of $B(p)$ quantizing bits per signal sample for the second group of subbands ($13 \leq p \leq 26$), which numbers $B(p)$ are related to the noise-masking curve of the human auditory system.” EX1008, 11:49-11:58. “[T]he predetermined allocation pattern $\{B(p)\}$ is loaded as a default value for the allocation pattern.” EX1008, 13:51-59.

218. Fifth, Veldhuis assigns additional bits to the audible subbands having $C(p;m)=1$ according to an iterative process based on which subband has the greatest “need of bits.” EX1008, 14:53-15:16, FIG. 9. The bits available for allocation in this step are those “saved” from the blocks having $C(p;m)=0$ (sum S). EX1008, 11:45-11:49, 12:24-31. By reallocating the saved bits in this way, Veldhuis’s system results “in an effective reduction of the audibility of quantizing noise during music passages presenting single tones.” EX1008, 13:4-6.

B. Overview of Galand

219. U.S. Patent No. 4,677,671 (“Galand”) discloses a method and device for coding a voice signal, which splits residual signals “over several sub-bands” and codes them. EX1009, Title, Abstract. Galand’s Figure 2 demonstrates this process:



EX1009, FIG. 2.

A sub-band coder 22 is “provided with filter set (FB) 30 which receives samples $X_{(n)}$ ” and “divides the pass band of the residual base-band signal into eight adjacent sub-bands 1-8.” EX1009, 4:33-37.

220. Relevant here, Galand teaches that bits are allocated based on several parameters, including the number of “useful sub-bands”—i.e., those that are available for bit allocation. EX1009, 4:64-5:6, 6:33-42. All other bands are not counted and thus do not have bits allocated to them.

221. In particular, Galand teaches that dynamic bit allocating device (DAB) 33 “dynamically determines for each block of processed samples, the bit rate $n_{(i)}$ to be allocated for requantizing the i^{th} sub-band samples,” using the following mathematical model for allocating bits, shown below:

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

EX1009, 4:59-5:5. This model includes the “[n]umber of useful sub-bands” (“p”) and requires summing over all “useful sub-bands” (e.g., $2 \leq j \leq 7$ in the equation above). EX1009, 4:61-5:9, 6:33-42.

C. A POSA would have been motivated to use Galand’s mathematical model in determining Veldhuis’s initial bit allocation.

222. A POSA would have been motivated to combine Veldhuis and Galand. In particular, a POSA would have been motivated to use Galand’s mathematical model for allocating bits in Veldhuis’s method. As I discussed above, both references describe methods and systems for subband coding of audio signals. EX1008, Abstract, 1:6-40, 2:14-21, 11:45-11:49, 11:67-12:4; EX1009, Abstract, 2:23-45, 3:11-31, 5:37-41, 6:33-42. Both references recognize, moreover, the importance of efficiently allocating bits to the encoded signal, with a focus on reducing quantizing noise. EX1008, 2:14-21; EX1009, 1:27-51. And, like Veldhuis, Galand discloses a method for allocating bits only to “useful sub-bands.”

EX1009, 6:33-42.

223. Veldhuis's method includes a first bit allocation step, in which bits are only allocated to those blocks that can be heard (i.e., those that are "counted").

EX1008, 11:37-12:14. Veldhuis teaches that the bit allocation pattern $\{B(p)\}$ is "related to the noise-masking curve of the human auditory system" and, in one embodiment, is calculated from " $L(p)$," which is the number of quantizing levels and is "related to th[e] noise-masking curve in a manner such that in each subband the signal-to-noise ratio is sufficiently high for not perceiving the quantizing noise." EX1008, 7:48-52, 11:49-58. But Veldhuis does not expressly disclose how the numbers of bits $B(p)$ is determined.

224. Thus, a POSA would have been motivated to look for these implementation details elsewhere. Galand provides such implementation details, including a mathematical model for allocating bits, shown below:

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

EX1009, 5:1-5.

225. A POSA would have been motivated to incorporate Galand's mathematical model for bit allocation into Veldhuis in order to determine the bit allocation for the available blocks. *See* Section X.D.5. Similar to Veldhuis, Galand's bit allocation is premised on assigning bits to only the "useful sub-bands"

(e.g., $2 \leq j \leq 7$ in the equation above). EX1009, 4:61-5:9, 6:33-42. Thus, Galand's equation provides a way to conveniently allocate bits in Veldhuis's system based on the number of "useful" bands that can be heard (i.e., that have a $C(p;m)$ value of 1). And, notably, implementing Galand's equation in Veldhuis would not change Veldhuis's second bit allocation step in any way. *See* Section X.D.6.

226. A POSA would have easily combined Veldhuis and Galand because both use conventional audio-processing hardware to implement their methods. *See, e.g.,* EX1008, 1:39-46, 5:38-43 ("The basic structure of such a system [Figure 1] is generally known."), 6:10-11 ("For the quantizing of the subband signals $x_p(k)$ known block-adaptive PCM methods are used."); EX1009, 3:4-10, 6:30-32 ("For more information about BCPCM coding and decoding, one should refer to the drawings of U.S. Pat. No. 4,216,354" (EX1036)), FIG. 1. Combining Veldhuis and Galand would have merely been a simple substitution of one known element (Galand's mathematical model for determining bit allocation) for another (Veldhuis's undisclosed implementation details for determining bit allocation) to obtain predictable results (allocating bits only to the blocks that can be heard, to improve the effect of noise). Thus, a POSA also would have had a reasonable expectation of success in making the combination.

D. Veldhuis in view of Galand renders claim 1 obvious.

- 1. Veldhuis discloses, or at least renders obvious, “[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:” [1P].**

227. To the extent that the preamble language is deemed limiting, Veldhuis discloses it, or at least renders it obvious.

228. Veldhuis describes a method for the “allocation of the numbers of quantizing bits $B(p;m)$ ” for a “digital audio signal.” EX1008, 2:22-24, 10:15-31. Veldhuis’s coder 1 includes filter bank 3, which “divides the music signal band” and “generates [the] subband signals $x_p(k)$ which are quantized blockwise.” EX1008, 5:53-66. As noted, the quantization is performed “blockwise,” meaning that the signal is divided into twenty-six subbands ($1 \leq p \leq 26$), which, in turn, are divided into “blocks” (denoted by $(p;m)$). EX1008, 2:67-3:2, 5:53-68, FIG. 5. Each block has a “block length” or “duration” (i.e., a temporal component) as well as “spectral components.” EX1008, 2:40-41, 3:32-49, 9:33-43.

229. Further, Veldhuis’s coder 1 has a buffer 6(p) to record the digital audio signals. EX1008, 6:12-18; *see also* EX1008, 10:65-68 (describing signal buffers 22). A POSA would have understood that the decomposing and buffering, occurring at filter bank 3 and buffer 6, results in “digital audio signals [that] are

obtained by buffering audio signals in frames and decomposing the digital signals into spectral components,” as required by the preamble. I further understand that the BPAI indicated that buffering is “either inherent or obvious to one of ordinary skill in the art.” EX1002, 253. I agree on this point, as I explained above. *See* pp. 58-59, n. 5.

230. Finally, “bit allocation means” 23 performs “[t]he allocation of the numbers of quantizing bits $B(p;m)$ per signal sample from the predetermined total number of B bits for an allocation window.” EX1008, 11:1-13. The complete allocation pattern $\{B(p;m)\}$ is then transferred to quantizers $8(p)$. EX1008, 12:60-62.

231. Veldhuis thus discloses, or at least renders obvious, all aspects of the preamble.

2. Veldhuis discloses “a step for dividing at least one frequency interval into a plurality of frequency-units;” [1A].

232. As I previously explained, Veldhuis’s coder 1 “divides the ... signal band of 0-22.05 kHz” “into a number of $p=26$ subbands.” EX1008, 5:53-59, FIG. 3. Veldhuis further samples each subband in “a time interval corresponding with one block of $M=32$.” EX1008, 9:67-10:13; *see also* EX1008, 10:26-31 (“The allocation window shown in FIG. 5 is divided into blocks, each block corresponding with 32 signal samples and also with a single associated value of the

characteristic parameter $G(p;m)$ and a single value of the number of quantizing bits $B(p;m)$ per signal sample, respectively.”), FIG. 5. Each of these blocks $(p;m)$ represents the claimed “frequency-unit.”

233. Only the blocks associated with subbands having band numbers $(13 \leq p \leq 26)$ are allocated bits using the method described in Veldhuis. These bands correspond to a frequency range of between 1,723 Hz and 22.05 kHz, as shown below:

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

Frequency
Interval
(1,723 Hz –
22.05 kHz)
Divided into
Plurality of
Frequency-
Units
($13 \leq p \leq 26$)

EX1008, FIG. 3 (annotated).

234. That range represents the claimed “frequency interval.” And, as I

previously explained, Veldhuis divides the interval into a plurality of frequency-units (i.e., blocks). EX1008, 5:53-68, 7:23-31. Accordingly, Veldhuis satisfies this limitation. I understand that, consistent with my opinion here, the BPAI found during prosecution that Veldhuis'463 satisfies this limitation. *See* EX1002, 253.

235. Even to the extent the limitation is step-plus-function, Veldhuis's disclosure is identical to the corresponding acts disclosed in the '757 patent. The '757 patent describes grouping transformed spectral samples into units, just as Veldhuis does. EX1001, 1:37-40, 4:21-24, FIG. 6; EX1008, 5:53-68, 7:23-31, 9:67-10:13, 10:26-31, FIG. 3, FIG. 5. Accordingly, in my view, Veldhuis discloses this limitation regardless of whether it is a step-plus-function limitation. At the very least, Veldhuis's description of dividing a frequency interval is equivalent to the corresponding acts disclosed in the '757 patent specification.

3. Veldhuis discloses “a step for obtaining a representative of each frequency-unit obtained;” [1B].

236. Veldhuis discloses this limitation. In particular, Veldhuis's level detector 7(p) determines “a characteristic parameter $G(p;m)$ representative of the signal level in th[e] block,” as I discussed above. EX1008, 6:14-18. “This characteristic parameter $G(p;m)$ is used for an optimal adjustment of an adaptive quantizer 8(p) for quantizing the signal block stored having block number m.” EX1008, 6:18-21. Accordingly, the parameter $G(p;m)$ is the claimed “representative.”

237. Indeed, Veldhuis explains that “the signal level can be represented” by $G(p;m)$ in the form of the average value or peak value of the amplitudes within a block. EX1008, 6:24-28. And, by determining the parameter $G(p;m)$, Veldhuis “obtains” it. Accordingly, Veldhuis satisfies this limitation. I understand that, consistent with my opinion here, the BPAI also found during prosecution that Veldhuis’463 satisfies this limitation. *See* EX1002, 254.

238. Even to the extent the limitation is step-plus-function, Veldhuis’s disclosure is identical to the corresponding acts disclosed in the ’757 patent. The ’757 patent describes that the “maximum spectral amplitude” is a “good worst case representative of the signals in the unit,” just as Veldhuis discloses the characteristic parameter $G(p;m)$ can be the peak value of the amplitudes within a block. EX1001, 3:55-59; EX1008, 6:14-28. Accordingly, in my view, Veldhuis discloses this limitation regardless of whether it is a step-plus-function limitation. At the very least, Veldhuis’s description of obtaining a representative is equivalent to the corresponding acts disclosed in the ’757 patent specification.

4. Veldhuis, alone or in combination with Galand, renders obvious “a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria;” [1C].

239. Veldhuis, alone or in combination with Galand, renders this limitation obvious.

240. Veldhuis categorizes the blocks within subbands $13 \leq p \leq 26$ into two

groups: (a) blocks that can be heard, and (b) blocks that cannot. EX1008, 11:23-46.

Veldhuis uses that information to determine the sum of saved bits S for allocating bits to the blocks that can be heard. EX1008, 14:7-23. This process satisfies claim 1's "counting" step, as I will now explain.

241. Veldhuis categorizes the blocks by comparing the characteristic parameter $G(p;m)$ (the claimed "representative") for each block to a threshold $T(p)$ that is "related to the thresholds of the human auditory system for just perceiving single tones in the respective frequency ranges" (the claimed "psychoacoustic criteria"). EX1008, 11:28-36. I understand that, consistent with my opinion here, the BPAI found during prosecution that the thresholds of Veldhuis'463 are "based on psychoacoustic criteria." *See* EX1002, 255. The result of Veldhuis's comparison, $C(p;m)$, determines which blocks are available for bit allocation.

242. For example, blocks "containing perceptually significant signal energy" (i.e., above the human auditory threshold), have their $C(p;m)$ values set to 1. EX1008, 2:49-59, 11:23-44. By contrast, blocks "containing no perceptually significant signal energy" (i.e., below the threshold) have their $C(p;m)$ values set to 0. EX1008, 2:49-59, 11:23-36. By determining which signal blocks can be heard and thus are available for bit allocation (i.e., those having $C(p;m)$ values equal to 1), Veldhuis "counts" such blocks. Indeed, the '757 patent describes its counting step in nearly identical terms. *See* EX1001, 4:46-48 (explaining that using a

“hearing threshold will eliminate units that are irrelevant according to psychoacoustic criteria” such that they will not be counted).

243. I understand that during prosecution of the '757 patent, the BPAI found that Veldhuis'463 does not disclose actively “counting” the “available frequency-units” (i.e., blocks that can be heard), but rather found that it instead discloses tracking a “count” of such units. *See* Section VI.C; EX1002, 256. However, in my opinion this is a distinction without a difference. In fact, as I previously noted, the Applicant acknowledged during prosecution that “there is no mention of a bandwidth counter” in the claims, and nothing in the claims requires actively counting by, for example, incrementing a counter. *See* EX1002, 225-26.

244. In any event, even if the BPAI's finding were correct, the BPAI failed to consider Veldhuis's entire disclosure. Veldhuis discloses that, when a particular block has a $C(p;m)$ value equal to 0, the number of available bits S for subsequent allocation is “increment[ed]” by the number of bits that would have otherwise been assigned to the block. EX1008, 13:51-14:23, FIG. 8. By incrementing S in this way, Veldhuis “counts” the bits associated with the unavailable blocks (i.e., those having $C(p;m)=“0”$). Veldhuis's express disclosure of counting bits for unavailable frequency bands would have motivated a POSA to employ a similar technique with respect to the available frequency bands. Doing so would have been nothing more than a mere design choice.

245. Further, a POSA would have found it obvious to actively count available frequency bands in light of Veldhuis's disclosure about incrementing a bit count associated with unavailable frequency bands (and subsequently made available to allocate to the available frequency bands). As I explained above, there are only two options for $C(p;m)$ (1 and 0). Given this limited choice, it would have been obvious to count the 1s (available blocks) instead of the 0s (unavailable blocks).

246. At the very least, affirmatively counting available frequency bands in Veldhuis would have been obvious to try. In Veldhuis, the total number of bits B available for allocation is known. *See, e.g.*, EX1008, 10:31-41. Further, there is only a limited number of choices as to how to determine the number of "saved" bits S : based on the blocks that can be heard or the blocks that cannot be heard. EX1008, 3:61-66, 9:67-10:10. In view of that limited choice, a POSA would have been motivated to try determining S by counting the number of blocks that can be heard.

247. A POSA would have likewise been motivated to affirmatively count available frequency bands in Veldhuis based on the disclosure of Galand, which teaches using the "number of useful sub-bands" as a parameter in the mathematical

model used for determining the bit allocation for a sub-band.⁶ EX1009, 4:61-5:9, 6:33-42. A POSA would have understood that, in order to determine the number of sub-bands that are “useful,” Galand counts them. Galand thus confirms what Veldhuis already teaches—i.e., that counting the units that exceed a threshold (i.e., those that can be heard, or those that are “useful”) was a desirable way to allocate bits. Accordingly, Veldhuis, or the combination of Veldhuis and Galand, renders this limitation obvious.

248. To the extent the limitation requires incrementing a counter, a POSA would have found it obvious to do so in the Veldhuis-Galand combination, because incrementing a counter was well known and commonly used in programming at the time of the '757 patent. *See, e.g.*, EX1038, 13:30-33 (“Counter 240 is a well known digital counter that counts to a maximum value, then overflows to a zero value, then counts back up to the maximum value.”); EX1042, 9:11-12 (“To encode, the number of MPSs in a run are counted by a simple counter.”). A POSA would have found it obvious to do so as a straightforward and preferred way to implement the Veldhuis-Galand combination’s counting in software.

249. Finally, the above analysis applies with equal force even to the extent

⁶ In the combination, Galand’s mathematical model is used for determining the initial bit allocation. *See* Sections X.C, X.D.5.

the limitation is step-plus-function. As explained above, it would have been obvious to implement the Veldhuis-Galand combination to count frequency-units by incrementing a counter, as disclosed in the '757 patent specification. EX1001, 4:29-48, 4:51-53, FIG. 1 (numeral 3), FIG. 2 (numerals 32-33), FIG. 3 (numeral 41); EX1008, 2:49-59, 3:61-66, 9:67-10:10, 10:31-41, 11:23-49, 13:51-14:23, FIG. 8; EX1009, 4:61-5:9, 6:33-42.

5. Veldhuis in view of Galand 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; and;” [1D].

250. The Veldhuis-Galand combination renders this limitation obvious. Veldhuis teaches a first bit allocation step in which bits $B(p)$ are allocated only to those blocks that can be heard (i.e., those having a $C(p;m)$ value equal to 1), while no bits are allocated to blocks that cannot be heard (i.e., those having a $C(p;m)$ equal to 0). EX1008, 11:44-55. Veldhuis further teaches that the “numbers $B(p)$ [of bit allocation pattern $\{B(p)\}$] are related to the noise-masking curve of the human auditory system” and, in one embodiment, are calculated as $B(p)=\log_2[L(p)]$. EX1008, 11:49-58. Here, “ $L(p)$ ” is the number of quantizing levels, which Veldhuis explains is “related to th[e] noise-masking curve in a manner such that in each subband the signal-to-noise ratio is sufficiently high for not perceiving the quantizing noise.” EX1008, 7:48-52. Although this $L(p)$ value may be

predetermined⁷ for each subband (*see, e.g.*, EX1008, FIG. 3), during Veldhuis's first bit allocation step, bits are only allocated to those blocks that can be heard (i.e., those that are "counted"). EX1008, 11:37-12:14.

251. Veldhuis does not expressly disclose specifically how the L(p) values (and thus the B(p) values) are determined. As discussed above, a POSA would have been motivated to look for these implementation details elsewhere. Galand provides such implementation details, including a mathematical model for allocating bits, shown below:

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

EX1009, 5:1-5.

⁷ I understand that the BPAI determined the initial bit allocations in Veldhuis are "predetermined" and thus do not teach this limitation. EX1002, 260. The distribution of values shown in Figure 3 of Veldhuis, however, is simply one example for a particular audio signal. A POSA would have understood that for different audio signals, a different initial bit allocation would result. In any event, in the Veldhuis-Galand combination relied upon here, bit allocation clearly is not predetermined, but rather determined using Galand's mathematical model.

This model includes both the “[n]umber of useful sub-bands” (“p”) and requires summing over all “useful sub-bands” (e.g., $2 \leq j \leq 7$ in the equation above). EX1009, 4:61-5:9, 6:33-42.

252. A POSA would have been motivated to incorporate Galand’s mathematical model for bit allocation into Veldhuis in order to determine the bit allocation for the available blocks. In combination with Veldhuis’s teachings to only allocate bits to the available blocks (i.e., corresponding to Galand’s “useful sub-bands”), using Galand’s bit allocation model would result in summing over the blocks for which $C(p;m)=1$ (i.e., the available blocks). The mathematical model of the Veldhuis-Galand combination mirrors Equation 1 of the ’757 patent, in which the model uses the number of available frequency units by likewise summing over “N,” the “Number of valid units.” A POSA would have thus understood that in the Veldhuis-Galand combination, the initial bit allocation is determined “using an approximate mathematical model using the counted number of available frequency-units,” as recited in claim 1.

253. Even to the extent the limitation is step-plus-function, the Veldhuis Galand combination’s bit-allocation using a mathematical model is equivalent to the corresponding acts disclosed in the ’757 patent specification. EX1001, 4:49-5:11, FIG. 1 (numeral 4), FIG. 3 (element 43); EX1008, 7:48-52, 11:37-12:14; EX1009, 4:61-5:9, 6:33-42. In both the Veldhuis-Galand combination and the ’757

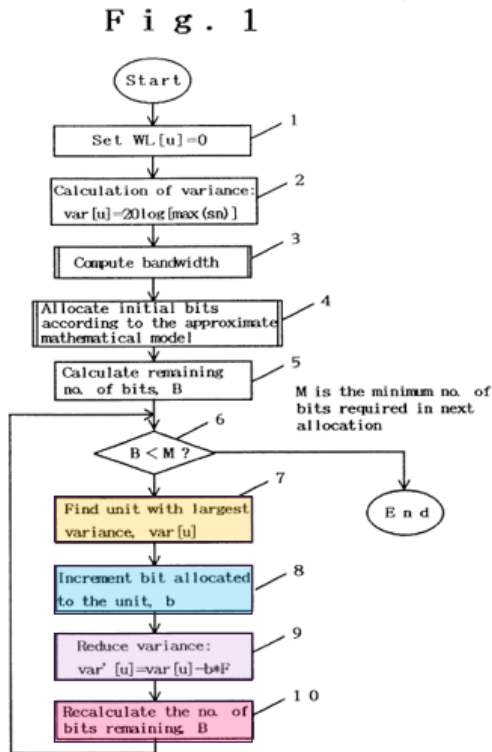
patent, the mathematical model allocates bits to the most energetic frequency bands and based on masking criteria. Thus, a POSA would have found the two models to be equivalent.

6. Veldhuis discloses, or at least renders obvious, “a second bit allocation step for determining further bit allocation for each of said available frequency-units based on psychoacoustic criteria.” [1E].

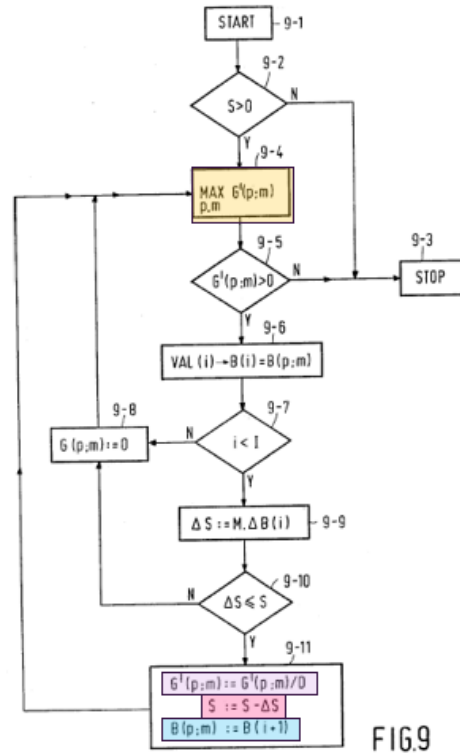
254. Veldhuis discloses, or at least renders obvious, this limitation. In particular, Veldhuis discloses a second bit allocation step in which the “sum S of saved numbers of bits” is allocated “to the blocks (p;m) within the allocation window having $C(p;m)=1$,” resulting in more bits being allocated to those blocks than initially allocated by the first allocation step. EX1008, 11:45-49, 12:24-31. In this way, Veldhuis necessarily determines further bit allocation based on psychoacoustic criteria because, in this second bit allocation step, additional bits are only assigned to those blocks that can be heard (i.e., blocks for which $C(p;m)=1$). And, as discussed above, the determination of which blocks can be heard in Veldhuis is made by comparing the characteristic parameter $G(p;m)$ for each block to a threshold $T(p)$ (the claimed “psychoacoustic criteria”) that is “related to the thresholds of the human auditory system for just perceiving single tones in the respective frequency ranges.” EX1008, 11:28-36; *see also* Section X.D.4.

255. Even beyond that, Veldhuis’s second bit allocation is implemented in

the same way as described in the '757 patent, as illustrated by a comparison of the '757 patent's Figure 1 and Veldhuis's Figure 9 below:



EX1001, FIG. 1 (annotated).



EX1008, FIG. 9 (annotated).

256. First, the “allocation commences at the block (p;m) having the largest value of parameter $G(p;m)$ ” (orange). EX1008, 12:30-36, 14:63-64, FIG. 9 (step 9-4). In this process, Veldhuis refers to $G'(p;m)$ as the “need of bits.” EX1008, 14:62-64. The initial value of $G'(p;m)$ is set as $G(p;m)$ for blocks having $C(p;m)=1$ and is set to 0 for blocks having $C(p;m)=0$. EX1008, 13:51-66. A POSA would have understood that this step 9-4 of Veldhuis corresponds to the '757 patent's step 7 of finding the unit having the largest representative (variance in the

embodiment of Figure 1).

257. Second, if a sufficient number of saved bits are available and other criteria are met (steps 9-5 to 9-10), the additional bits are allocated to the block (blue). EX1008, 12:32-35, 15:14-15, FIG. 9 (step 9-11 (“ $B(p;m) := B(i+1)$ ”). A POSA would have understood that this corresponds to the ’757 patent’s step 8 of incrementing bits allocated to the unit.

258. Third, Veldhuis’s sum S is reduced by the number of bits allocated to the block, thus indicating the number of saved bits remaining (pink). EX1008, 12:36-39, 15:12-13, FIG. 9 (step 9-11 (“ $S := S - \Delta S$ ”). A POSA would have understood that this corresponds to the ’757 patent’s step 10 of recalculating the number of bits remaining.

259. Fourth, the parameter $G'(p;m)$ is reduced for the block so as “to evenly distribute the sum S of the saved numbers of bits over the blocks $(p;m)$ having $C(p;m) = '1'$ ” (purple). EX1008, 12:39-44, 15:10-11 (step 9-11 (“ $G'(p;m) := G'(p;m)/D$ ”). A POSA would have understood that this corresponds to the ’757 patent’s step 9 of reducing the variance.

260. A POSA would have understood that the order in which the ’757 patent’s steps 8-10 are performed would not affect the result because they are simply adjusting various values before repeating the step of choosing the unit with the largest variance and continuing the process.

261. Veldhuis explains that this iterative process “exploit[s] the psychoacoustic effect of temporal masking, which means the property of the human auditory system that its threshold for perceiving signals shortly before and shortly after the occurrence of another signal which has a relatively high signal energy appears to be temporarily higher than in the absence of the latter signal.” EX1008, 4:34-40. And, indeed, the effect is expressly captured by Veldhuis’s coefficient “D” (shown in step 9-11 of Figure 9). I understand that, during prosecution, the BPAI found that the D value disclosed in Veldhuis’463 is not based on psychoacoustic criteria. EX1002, 258. I disagree. The similarity between how Veldhuis describes its D value and how the ’757 patent describes its “psychoacoustic weighting factor F” makes clear that Veldhuis’s D value is in fact based on psychoacoustic criteria as claimed.

262. In the ’757 patent, once bits are allocated to a unit in the second bit allocation step, the variance for that unit is reduced by a factor of F so that remaining bits are not disproportionately allocated to the same unit in the next pass, as shown in step 9 of Figure 1. EX1001, 4:11-13. Although the ’757 patent refers to F as the “psychoacoustic weighting factor,” no specific method for determining F is disclosed; rather, F is merely set to a value that the inventors have determined to be appropriate based on the results of setting F to various values. EX1001, 5:19-33. In particular, the patent explains that F is set to whatever value

will result in an error spectrum that most closely follows the original signal, but ultimately it is simply a coefficient that is adjusted to obtain desired characteristics. EX1001, 5:12-24.

263. Similarly, in Veldhuis, the D value is used to “reduce[]” the value of $G(p;m)$ “so as to evenly distribute the number of quantizing bits” in subsequent passes. EX1008, 15:54-60; *see also* EX1008, 14:57-15:16. And, like the ’757 patent, Veldhuis explains that the D value “can be determined such that the improvement in the signal-to-noise ratio due to the increment of the number of quantizing bits per signal sample with an amount [that] is approximately reflected in the reduction of the need of bits $G'(p;m)$.” EX1008, 15:60-65. Further, Veldhuis suggests that D can be a function of the bands, which the BPAI did not appear to consider. EX1008, 15:65-16:6. In any event, both F in the ’757 patent and D in Veldhuis serve as coefficients that decrease the representative value with each iteration in order to ensure even distribution in the second bit allocation step.

264. Based on this similarity, a POSA would have understood that Veldhuis’s D value reflects psychoacoustic criteria. And, indeed, the BPAI itself noted that the phrase “based on psychoacoustic criteria” can be “interpreted broadly as very indirectly based on some psychoacoustic criterion.” EX1002, 257. At the very least, Veldhuis discloses such an indirect relationship.

265. Even to the extent the limitation is step-plus-function, Veldhuis’s

second bit allocation step is equivalent to the corresponding acts disclosed in the '757 patent specification. EX1001, 3:63-4:18, FIG. 1 (numerals 6-10); EX1008, 4:34-40, 11:44-49, 12:24-44, 13:51-66, 14:57-15:16, 15:54-16:6, FIG. 9. As I demonstrated above, both Veldhuis and the '757 patent iteratively allocate bits based on psychoacoustic criteria. Thus, in my view, the two are equivalent.

E. Veldhuis in view of Galand renders claim 2 obvious.

266. The combination of Veldhuis and Galand renders obvious a “[m]ethod 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,” as recited in claim 2.

267. As I previously explained with respect to limitation [1D], the Veldhuis-Galand combination discloses allocating bits ($B(p)$) to blocks ($p;m$) using an approximate mathematical model. *See* Section X.D.5. Moreover, it would have been obvious to derive that model from an in-band masking effect.

268. Veldhuis, for example, teaches that bit allocation numbers $B(p)$ are “related to the thresholds for masking noise in the critical bands of the human auditory system by single tones in the centre of the respective critical bands.” EX1008, 2:63-66; *see* EX1008, 7:48-52, 11:53-56. Veldhuis further explains that “quantizing noise will be optimally masked by the signals within [a] subband when the quantizing takes account of the noise-masking curve of the human auditory

system (this curve indicates the threshold for masking the noise in a critical band by a single tone in the centre of the critical band).” EX1008, 1:52-58.

269. Veldhuis recognizes that “[t]he refined quantization [of the invention] during music passages presenting single tones ... results in an effective reduction of the audibility of quantizing noise without the need of increasing the average number of quantizing bits per output signal sample of the coder.” EX1008, 3:66-4:2; *see also* EX1008, 13:4-39. Galand likewise recognizes the importance of reducing quantizing noise. EX1009, 1:27-35.

270. A POSA would have understood that a reduction in the audibility of quantizing noise, as taught both by Veldhuis and Galand, results when “an error spectrum ... follow[s] more closely to the original signal spectrum,” as claimed.

271. Thus, Veldhuis in view of Galand renders claim 2 obvious.

F. Veldhuis in view of Galand renders claims 3, 7, 11, and 14 obvious.

272. The combination of Veldhuis and Galand renders obvious a “[m]ethod according to claim 2, wherein each frequency-unit is represented by a single spectral component,” as recited in claim 3. The combination also renders obvious claims 7, 11, and 14, which similarly recite each frequency-unit being represented by a single spectral component. *See* EX1001, Claims 7, 11, 14. Claim 7 depends from claim 6, which depends from claim 1. Claim 11 depends from claim 10, which also depends from claim 1. And claim 14 depends directly from claim 1.

273. As I explained above, Veldhuis teaches “a characteristic parameter $G(p;m)$ ” or “the signal level in th[e] block” as a “representative value” of claim 1. EX1008, 6:12-34. Veldhuis teaches, moreover, that “the signal level can be represented ... by the peak value of the amplitude of the signale [sic] samples in a block.” EX1008, 6:12-34. The peak value, in turn, is simply the amplitude of the single spectral component within the block having the largest magnitude.

274. Alternatively, Veldhuis recognizes that “the quantizing noise is especially audible in music passages presenting single tones. During such music passages the greater part of the subbands have very little or no signal energy from the mid frequency portion of the music signal band onwards, whereas only a single spectral component having significant signal energy occurs in each of the few remaining subbands.” EX1008, 8:68-9:7. In this instance, Veldhuis teaches that each frequency-unit is represented by a single spectral component. Accordingly, Veldhuis discloses each frequency-unit being represented by a single spectral component as required by the claims.

275. Thus, Veldhuis in view of Galand renders claims 3, 7, 11, and 14 obvious.

G. Veldhuis in view of Galand renders claim 6 obvious.

276. The combination of Veldhuis and Galand renders obvious 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,” as recited in claim 6.

277. As I previously explained, the '757 patent does not meaningfully discuss or define a “psychoacoustically weighted inverse relationship” with the representative of each frequency-unit. *See* Section VIII.G. But, as also explained above, adverse psychoacoustic effects are reduced through the iterative bit allocation process shown in Figure 1 of the '757 patent such that the bit allocation is determined based on a psychoacoustically weighted inverse relationship with the variance of each frequency-unit. *See* Section VIII.G.

278. That is what Veldhuis discloses. *See* Sections X.D.4-6. Veldhuis's second bit allocation step assigns bits such that more bits are allocated to blocks “having the largest value of parameter $G(p;m)$ ” (the claimed “representatives”), and no bits are assigned to blocks that cannot be heard as determined based on a threshold $T(p)$ “related to the thresholds of the human auditory system for just perceiving single tones in the respective frequency ranges” (the claimed “psychoacoustic criteria”). *See* EX1008, 11:28-36, 13:51-16:17. As Veldhuis explains, this method reduces “the audibility of quantizing noise in the replica of the audio signals ... in an effective manner without increasing the average number of bits per sample of the quantized output signals of the coder.” EX1008, 2:14-20; *see* EX1008, 4:33-52, 13:4-6. And, like the '757 patent, Veldhuis iteratively

assigns bits in this manner until bits are no longer available for allocation. EX1008, 14:53-16:18, FIG. 9. Thus, Veldhuis's adaptive bit allocation performs the same band identification and iterative bit assignment disclosed in the '757 patent itself.

279. Accordingly, Veldhuis in view of Galand renders claim 6 obvious.

H. Veldhuis in view of Galand renders claim 10 obvious.

280. The combination of Veldhuis and Galand renders obvious 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," as recited in claim 10.

281. As I previously explained with respect to limitations [1B] and [1C], Veldhuis determines which blocks (p;m) should receive bits based on whether the blocks can be heard. *See* Sections X.D.3-4. This determination is based on a threshold $T(p)$ "related to the thresholds of the human auditory system for just perceiving single tones in the respective frequency ranges." EX1008, 11:28-36.

282. By accounting for blocks that are above the threshold, Veldhuis counts blocks that exceed a human hearing threshold, as required by the claim. Alternatively, in view of Galand's disclosure of using the "[n]umber of useful sub-bands" (i.e., available frequency-units), Veldhuis at least renders obvious counting blocks that exceed a human hearing threshold. Once again, Veldhuis's disclosure is nearly identical to the '757 patent's. *See* EX1001, 4:46-48 (explaining that using a

“hearing threshold will eliminate units that are irrelevant according to psychoacoustic criteria”).

283. Thus, Veldhuis in view of Galand renders claim 10 obvious.

I. Veldhuis in view of Galand renders claim 16 obvious.

284. The combination of Veldhuis and Galand renders obvious a “method according to claim 1, wherein each frequency-unit is obtained by dividing said spectral and temporal structure with at least one frequency interval,” as recited in claim 16.

285. As I explained above with respect to limitations [1P] and [1A], Veldhuis discloses bit allocation for an audio signal having spectral and temporal structure. *See* Sections X.D.1-2. Veldhuis divides the signal having a frequency range of between 1,723 Hz and 22.05 kHz (the claimed “frequency interval”) into blocks (p;m) (thus exploiting the signal’s temporal structure). EX1008, 4:53-68.

286. In block (p;m), “p” represents the subband and “m” represents the block. EX1008, 5:53-68. The subbands are frequency bands, as shown in Figure 3:

p	f _{co} [H z]		W(p) [H z]	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

Frequency Interval (1,723 Hz – 22.05 kHz) Divided into Plurality of Frequency-Units (13 ≤ p ≤ 26)

EX1008, FIG. 3 (annotated).

EX1008, 1:53-59, 5:1-3, 6:35-42. By dividing the spectrum into subbands, Veldhuis obtains the frequency-units, as required by the claim.

287. Thus, Veldhuis in view of Galand renders claim 16 obvious.

XI. GROUND 4: CLAIMS 1-17 OF THE '757 PATENT ARE OBVIOUS OVER THE COMBINATION OF VELDHUIS, GALAND, AND REAVES.

288. As I explain below, it is my opinion that claims 1-17 of the '757 patent are obvious over the combination of Veldhuis, Galand, and Reaves.

A. Overview of Reaves

289. U.S. Patent No. 5,579,431 ("Reaves") discloses speech detection in

the presence of noise by determining variance over time of frequency band limited energy. Specifically, Reaves describes a speech-recognition device for detecting “the start and end of a segment containing speech within an input audio signal which contains both speech segments and nonspeech noise or background segments.” EX1010, Title, 1:9-13. Reaves’s device uses energy values in particular frequency bands to calculate “the beginning and the ending points of speech.” EX1010, 5:66-6:3.

290. Specifically, “[t]he variance of the frequency band limited energy” is calculated. EX1010, 4:42-43. Reaves explains that the variance quantifies the fluctuation of the frequency band limited energy. EX1010, 2:3-13. This “variance V” is compared with a “threshold,” and when the variance is above the threshold, Reaves determines that the “beginning of speech” has occurred. EX1010, 5:21-6:7, FIG. 5.

B. A POSA would have combined Reaves with Veldhuis and Galand.

291. In the combination proposed here, Reaves is used to determine a variance of a spectral component as a representative in place of the parameter $G(p;m)$ taught by Veldhuis. A POSA would have been motivated to use Reaves in the Veldhuis-Galand combination.

292. As I previously explained with respect to limitation [1B], Veldhuis uses characteristic parameters $G(p;m)$ as a representative of blocks $(p;m)$,

comparing $G(p;m)$ to threshold $T(p)$ to determine whether each block $(p;m)$ contains “perceptually significant signal energy.” See Section X.D.3; EX1008, 11:35-11:44. In a similar manner, Reaves teaches that its speech-recognition device compares the variance of frequency band-limited energy of the signal in a single frequency band to a threshold to determine whether the input signal contains a “beginning of speech.” EX1010, 4:17-28, 5:25-30. A POSA would have understood that detecting whether there is speech is related to the perceptibility of the signal, i.e., as compared to the background noise. EX1010, 1:33-37, 5:23-24, FIG. 5.

293. In other words, Veldhuis and Reaves are similar in that they use a given representative value—parameter $G(p;m)$ in Veldhuis and “variance” in Reaves—to detect perceptually significant signal energy. The variance in Reaves is of the “frequency band limited energy,” which is described as “[a]ny quantity that varies generally monotonically with the energy of the frequency band limited signal,” and may be, for example, “the peak of the value of the frequency band limited signal.” EX1010, 4:17-63. Similarly, Veldhuis teaches the signal level can be represented by the peak value of the amplitude of the signal samples. EX1008, 6:24-28.

294. Reaves teaches benefits to using a variance in detecting perceptually significant signal energy. Specifically, “[t]he use of the variance allows detection

which is relatively independent of the absolute signal-to-noise ratio with the signal and allows accurate detection within a wide variety of backgrounds such as music, motor noise, and background noise, such as other speakers.” EX1010, Abstract. In other words, using a variance allows for more effective detection of the desired sounds “above a ‘noise floor’ of relatively low fluctuation.” EX1010, 1:51-53, 2:3-14. Based on these teachings, a POSA would have been motivated to use a variance as taught in Reaves in place of the parameter $G(p;m)$ taught in Veldhuis.

295. A POSA would have known how to combine Reaves with the Veldhuis-Galand combination and would have had an expectation of success in forming the combination. Specifically, as noted, the combination would use a variance, as taught in Reaves, instead of parameter $G(p;m)$ (i.e., the “average value of the amplitude ... of the signal samples” or the “peak value of the amplitude of the signal samples”), as taught in Veldhuis. EX1008, 6:24-28, 11:35-11:44; EX1010, 4:42-43. Like Veldhuis and Galand, Reaves also uses conventional audio processing hardware to implement the methods thereof. EX1010, Abstract (“The device can be easily implemented using off-the-shelf hardware ...”). A POSA would have easily incorporated Reaves’s algorithm for calculating variance into the Veldhuis-Galand combination in view of the conventional components described in the references.

C. Veldhuis in view of Galand and Reaves renders claims 4, 8, 12, and 15 obvious.

296. The combination of Veldhuis, Galand, and Reaves renders obvious a “[m]ethod 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,” as recited in claim 4. The combination also renders obvious claims 8, 12, and 15, which similarly recite the representative being a variance calculated from an amplitude of a maximum spectral component of each frequency-unit. *See* EX1001, Claims 8, 12, 15. Claim 8 depends from claim 6, which depends from claim 1. Claim 12 depends from claim 10, which likewise depends from claim 1. Claim 15 depends directly from claim 1.

297. As I explained above with respect to limitation [1B], Veldhuis teaches that “characteristic parameter $G(p;m)$ ” represents “the signal level in this block” (the claimed “representative value” of claim 1). *See* Section X.D.3. Veldhuis further teaches that “the signal level can be represented ... by the peak value of the amplitude of the signal samples in a block.” EX1008, 6:24-28.

298. As I discussed above, the ’757 patent takes a broad view of “variance.” Section IX.C. In any event, Reaves expressly teaches using a variance of a signal energy level to determine the perceptibility of the sound. In the combination with Reaves, instead of using the peak value of the amplitude as taught in Veldhuis, the claimed “representative value” is determined as a variance

thereof. *See* Section XI.B. Reaves expressly recognizes that its “frequency band limited energy,” from which the variance is determined, can be the “peak of the value” of the signal. EX1010, 4:17-28. Thus, the variance is calculated from peak values of the amplitude, or as a POSA would recognize is the same, an amplitude of a maximum spectral component, as claimed.

D. Veldhuis in view of Galand and Reaves renders claims 5, 9, 13, and 17 obvious.

299. The combination of Veldhuis, Galand, and Reaves renders obvious a “[m]ethod according to claim 2, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit,” as recited in claim 5. The combination also renders obvious claims 9, 13, and 17, which similarly recite the representative being a variance calculated from an average of a spectral component of each frequency-unit. *See* EX1001, Claims 9, 13, 17. Claim 9 depends from claim 6, which depends from claim 1. Claim 13 depends from claim 10, which likewise depends from claim 1. Claim 17 depends directly from claim 1.

300. As I explained above with respect to limitation [1B], Veldhuis teaches that “characteristic parameter $G(p;m)$ ” represents “the signal level in this block” (the claimed “representative value” of claim 1). *See* Section X.D.3. Veldhuis further teaches that “the signal level can be represented by the average value of the amplitude or the power of the signal samples of a block.” EX1008, 6:24-28.

301. Again, Reaves teaches using a variance of a signal energy level to determine the perceptibility of the sound. In the combination with Reaves, instead of using the average value of the amplitude as taught in Veldhuis, the “representative value” is determined as a variance thereof. *See* Section XI.B. Based on Reaves’s teaching of determining a variance based on the “peak of the value,” and on Veldhuis’s teachings of using either the “peak” or “average” in its method, it would have been obvious to a POSA to determine the variance of the average value of the amplitude. Thus, the variance is calculated from the average value of the amplitude, or as a POSA would recognize is the same, an average value of a spectral component, as claimed.

E. Veldhuis in view of Galand and Reaves renders claims 1-3, 6-7, 10-11, 14, and 16 obvious

302. Given that Reaves discloses the representative recited in claims 4 (“a variance which is calculated from an amplitude of a maximum spectral component”) and 5 (“a variance which is calculated from an average of a spectral component”), it also discloses the representative recited in independent claim 1. At the very least, Reaves discloses determining the peak and mean amplitude values within each band, which indisputably would qualify as a “representative”—i.e., “a characteristic representative of the signal.” EX1002, 254 (cleaned up).

303. Thus, even if the Board were to conclude that the combination of Veldhuis and Galand does not render claim 1 obvious, the combination of

Declaration of Dr. Tim A. Williams
U.S. Patent No. 6,339,757

Veldhuis, Galand, and Reaves does. Accordingly, Veldhuis, Galand, and Reaves
render claims 1-3, 6-7, 10-11, 14, and 16 obvious as well.

XII. CONCLUSION

In signing this declaration, I recognize that the declaration will be filed as evidence in a contested case before the Patent Trial and Appeal Board of the United States Patent and Trademark Office. 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.

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 so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code.

Executed this 6th day of September 2024 in Danville, California.

Respectfully submitted,



Tim A. Williams, Ph.D.