FILE HISTORY

US 6,339,757

PATENT:	6,339,757
INVENTORS:	TEH DO HUI TAN PEK YEW NEO SUA HONG
TITLE:	Bit allocation method for digital audio signals
APPLICATION NO:	US1993161798A
FILED:	06 DEC 1993
ISSUED:	15 JAN 2002
COMPILED:	16 SEP 2021

Roku, Inc. EX1002
U.S. Patent No. 6,339,757

ISSUE CLASSIFICATION	Class	Subclass
	704	229

6339757

UTILITY SERIAL NUMBER	PATENT DATE	PATENT NUMBER
08/161738	JAN 1 5 2002	6339757

SERIAL NUMBER	FILING DATE	CLASS	SUBCLASS	GROUP ART UNIT	EX
08/161,798	12/06/93	~~341~~ 704	229	2641 2648	4

APPLICANTS: DO HUI TEH, SINGAPORE, SINGAPORE; PEK YEW TAN, SINGAPORE, SINGAPORE; SUA HONG NEO, SINGAPORE, SINGAPORE,

CONTINUING DATA********
VERIFIED

none, IC

sll 08/161796 161797

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FOREIGN/PCT APPLICATIONS********
VERIFIED JAPAN 5-030712 02/19/93

yes, IC

Foreign priority claimed / 35 USC 119 conditions met / Verified and Acknowledged	AS FILED	STATE OR COUNTRY	SHEETS DRWGS.	TOTAL CLAIMS	INDEP. CLAIMS	FILING FEE RECEIVED	ATTORNEY'S DOCKET NO.
☑ yes ☐ no / ☑ yes ☐ no / IC, 6/10/96 Examiner's initials	→	SGX	8	4	1	$710.00	0348683

BEVERIDGE, DEGRANDI, WEILACHER & YOUNG
1850 M STREET, N. W., SUITE 800
WASHINGTON, DC 20036

TITLE: BIT ALLOCATION METHOD FOR ~~TRANSFORM CODER~~ DIGITAL AUDIO SIGNALS

U.S. DEPT. of COMM.-Pat. & TM Office— PTO-436L (rev. 10-78)

10-4-01

PARTS OF APPLICATION FILED SEPARATELY

M. Jones
Applications Examiner

NOTICE OF ALLOWANCE MAILED		CLAIMS ALLOWED	
9.28.01	MICHAEL N. OPSASNICK, Assistant Examiner	Total Claims: 2021	Print Claim: 1

ISSUE FEE (W)		DRAWING		
Amount Due	Date Paid	Sheets Drwg.	Figs. Drwg.	Print Fig.
$1240.00	12-3-01	8	8	3

FAN TSANG
SUPERVISORY PATENT EXAMINER
TECHNOLOGY CENTER 2600
Primary Examiner

ISSUE BATCH NUMBER: 473

Label Area

PREPARED FOR ISSUE

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Form PTO-436A (Rev. 8/92) Final Drawings (sheet set

ISSUE FEE IN FILE
Issue Fee

6,339,757

BIT ALLOCATION METHOD FOR DIGITAL AUDIO SIGNALS

Transaction History

Date	Transaction Description
12-06-1993	Request for Foreign Priority (Priority Papers May Be Included)
01-10-1994	Application Captured on Microfilm
01-25-1994	Transfer Inquiry
02-22-1994	Case Docketed to Examiner in GAU
10-02-1995	Information Disclosure Statement (IDS) Filed
10-02-1995	Information Disclosure Statement (IDS) Filed
05-16-1996	Preexamination Location Change
06-10-1996	Non-Final Rejection
06-17-1996	Mail Non-Final Rejection
10-17-1996	Response after Non-Final Action
10-17-1996	Incoming Letter Pertaining to the Drawings
10-17-1996	Request for Extension of Time - Granted
11-01-1996	Date Forwarded to Examiner
01-02-1997	Final Rejection
01-06-1997	Mail Final Rejection (PTOL - 326)
06-06-1997	Response after Final Action
06-06-1997	Request for Extension of Time - Granted
07-02-1997	Advisory Action (PTOL-303)
07-02-1997	Date Forwarded to Examiner
07-07-1997	Notice of Appeal Filed
07-07-1997	Request for Extension of Time - Granted
07-08-1997	Mail Advisory Action (PTOL - 303)
09-08-1997	Appeal Brief Filed
09-26-1997	Date Forwarded to Examiner
12-08-1997	Examiner's Answer to Appeal Brief
12-09-1997	Mail Examiner's Answer
02-09-1998	Request for Oral Hearing
02-09-1998	Reply Brief Filed
02-24-1998	Date Forwarded to Examiner
04-22-1998	Mail Reply Brief Noted by Examiner
04-22-1998	Reply Brief Noted by Examiner
04-23-1998	Mail Advisory of Non-Entry of Reply Brief
04-23-1998	Advisory on Non-Entry of Reply Brief
04-21-1999	Request for Oral Hearing
11-24-1999	PTAB Remand to the Examiner
02-01-2000	Mail Miscellaneous Communication to Applicant
02-01-2000	Miscellaneous Communication to Applicant - No Action Count
06-08-2000	Notification of Appeal Hearing
10-06-2000	Case Docketed to Examiner in GAU
07-30-2001	Mail PTAB Decision on Appeal - Reversed
07-30-2001	PTAB Decision - Examiner Reversed
09-28-2001	Mail Notice of Allowance
09-28-2001	Notice of Allowance Data Verification Completed
10-04-2001	Receipt into Pubs
10-04-2001	Receipt into Pubs
11-07-2001	Workflow - File Sent to Contractor
12-03-2001	Issue Fee Payment Verified
12-03-2001	Workflow - Drawings Finished
12-03-2001	Workflow - Drawings Matched with File at Contractor
12-03-2001	Workflow - Drawings Received at Contractor
12-03-2001	Workflow - Drawings Sent to Contractor
12-03-2001	Issue Fee Payment Received
12-07-2001	Receipt into Pubs

12-12-2001	Application Is Considered Ready for Issue
12-14-2001	Receipt into Pubs
12-28-2001	Issue Notification Mailed
01-15-2002	Recordation of Patent Grant Mailed
01-15-2002	Patent Issue Date Used in PTA Calculation

08/161,798

[redacted]

APPROVED FOR LICENSE [ ]

INITIALS _____

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CONTENTS

Date Entered or Counted			Date Received or Mailed
	1.	Application Prints papers.	
	2.	Priority Paper	12/6/94
	3.	PTO-1449 ART	10-2-95
	4.	Corrected Filing Receipt	8-17-95
6/12	5.	Res (3)	6-17-96
	6.	EXT. OF TIME (1 month)	10-17-96
	7.	AMDT. A + DRAFTSMANS LETTER	10-17-96
1/2	8.	Final Rej (3)	4-6-97 JR
	9.	Ext of Time (2)	6-6-97
	10.	Amdt B (RCE)	06-06-97
	11.	Advisory Action	7/8/97
	12.	EXT. OF TIME (3 month)	07-07-97
	13.	NOTICE OF APPEAL 99-1793	07-07-97
	14.	APPEAL BRIEF	09-08-97
12/8	15.	Examiner Answer	12/9/97
	16.	REPLY BRIEF	2-9-98
	17.	ORAL Hearing	2-9-98
4/22	18.	Att. Reply to Reply Brief	4/23/98
	19.	Information Disclosure Statement	10/25/99
	20.	Remand	11-24-99
	21.	Letter	2-1-00
	22.	Hearing (2-1-00 TC)	6/8/00
	23.	Waiver	5/22/01
	24.	Board Decision - REVERSED	7-30-01
9-28	25.	Notice of Allowance	9-28-01
12/12/01	26.	8 1	12/3/01
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STAPLE — AREA

☆U.S.GOVERNMENT PRINTING OFFICE 1997-430-220

PATENT NUMBER

ORIGINAL CLASSIFICATION

CLASS	SUBCLASS
704	219

APPLICATION SERIAL NUMBER
09/161798

CROSS REFERENCE(S)

CLASS	SUBCLASS (ONE SUBCLASS PER BLOCK)			
704	229	220		

APPLICANT'S NAME (PLEASE PRINT)
TEH ET AL

IF REISSUE, ORIGINAL PATENT NUMBER

INTERNATIONAL CLASSIFICATION

G	1	0	L	.	19	/04

GROUP ART UNIT
2645

ASSISTANT EXAMINER (PLEASE STAMP OR PRINT FULL NAME)
MICHAEL N. OPSASNICK

PRIMARY EXAMINER (PLEASE STAMP OR PRINT FULL NAME)
FAN TSANG

PTO 270
(REV. 5-91)

ISSUE CLASSIFICATION SLIP

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PATENT AND TRADEMARK OFFICE

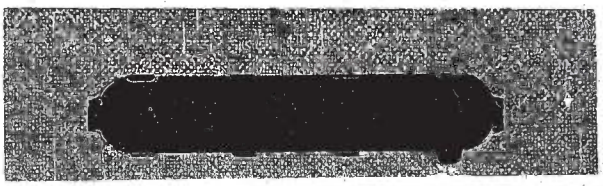
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SYMBOLS

✓	Rejected
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N	Non-elected
I	Interference
A	Appeal
O	Objected

Final	Original
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SEARCHED

Class	Sub.	Date	Exmr.
395	2.38	6/10/96	IC
395	2.39	6/10/96	IC
381	29	6/10/96	IC
	30	6/10/96	IC
updated		12/27/96	JMG

SEARCH NOTES

	Date	Exmr.
APS Search (see enclosed)	6/10/96	IC
Maya Search (see enclosed)	6/10/96	IC
updated APS MAYA	12/27/96	JMG

INTERFERENCE SEARCHED

Class	Sub.	Date	Exmr.
704	229	9/25/01	MC

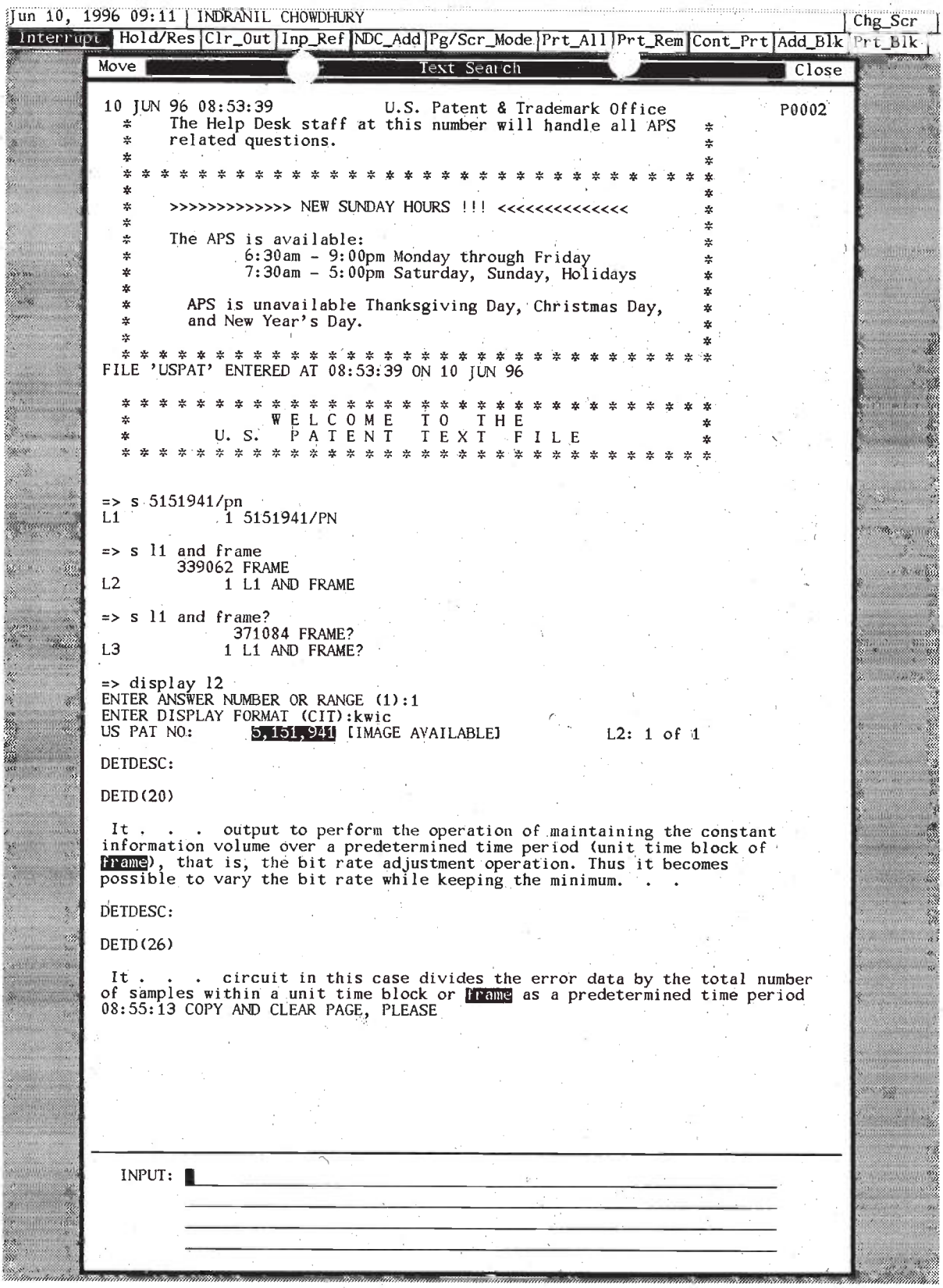
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ProQuest Search 6/9/96

IEEE/IEE Publications Ondisc Jan 1990 - Mar 1996

Search Options:
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Num	Search	Hits
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L3 1 L1 AND FRAME?
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US PAT NO: 5,151,941 [IMAGE AVAILABLE] L2: 1 of 1
DETDESC:
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It . . . output to perform the operation of maintaining the constant information volume over a predetermined time period (unit time block of frame), that is, the bit rate adjustment operation. Thus it becomes possible to vary the bit rate while keeping the minimum. . .
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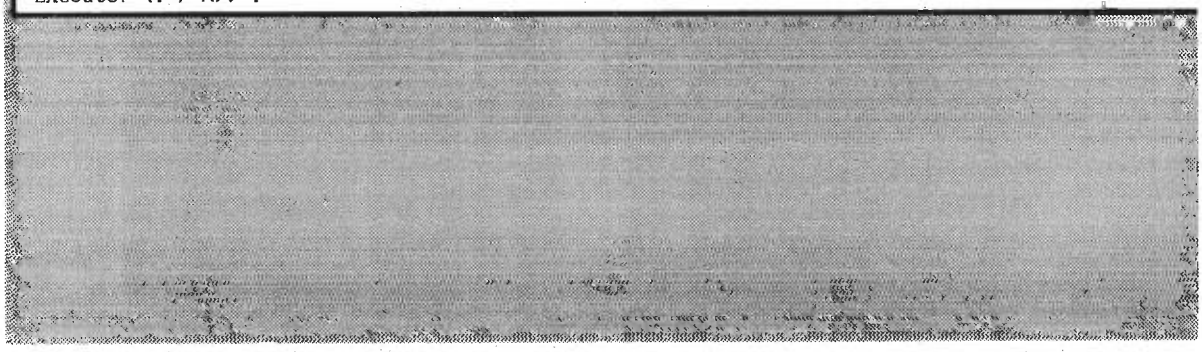
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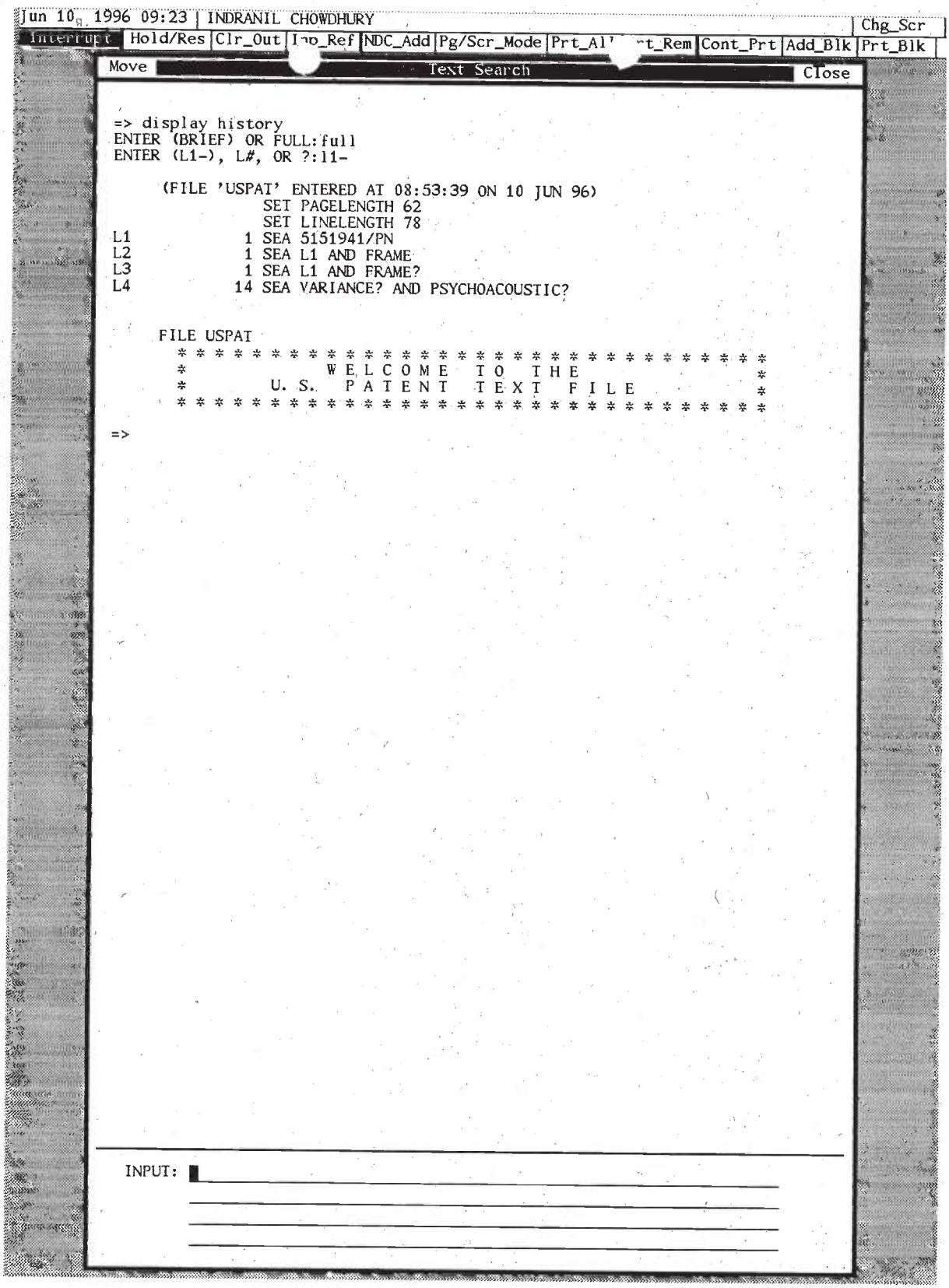
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L2 1 SEA L1 AND FRAME
L3 1 SEA L1 AND FRAME?
L4 14 SEA VARIANCE? AND PSYCHOACOUSTIC?
FILE USPAT
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Maya Search.

Ref	Patent Id	Issue/File	Class	Title
1	05151941	Sep 29 1992 Sep 7 1990	381/46	Digital signal encoding apparatus

Inventor: Nishiguchi; Masayuki et al.
Assignee: Sony Corporation
Abstract:

A digital signal encoding apparatus in which input digital signals are divided into a plurality of frequency bands so that the bandwidth of each frequency band will be greater the higher the frequency band, the allowable noise level is set for each frequency band on the basis of the energy value of each frequency band, and components of each frequency band are quantized with the number of bits consistent with the level difference between the energy of each frequency band and the preset allowable noise level. The output information volume following the quantization is detected and the number of bits of allocation for quantization is corrected in dependence upon the error between the detection output and the target value to render the information volume constant over a predetermined time period to enable bit rate adjustment or bit packing with lesser signal deterioration by a simplified construction.

Exemplary Claim [1]:

1. A digital signal encoding apparatus comprising

noise level setting means for dividing input digital signals into a plurality of frequency bands so that the bandwidth of each frequency band is broader as the frequency of the frequency bands increases, said noise level setting means setting the allowable noise level of each of the plurality of frequency bands on the basis of the energy of each

said
frequency band,
quantization means for quantizing components of each frequency band
with a number of bits consistent with the level difference between the
energy of each frequency band and that of the output of said noise level
setting means, and
means for detecting the number of bits allocated to the output
information of said quantization means and for adjusting the number of
bits of the output information over a predetermined time interval to be
constant as a function of the detection output.

2 05142656 Aug 25 1992 381/37 Low bit rate transform coder,
Nov 4 1991 decoder, and encoder/decoder for
high-quality audio
Status: certificate of correction has been issued
Inventor: Fielder; Louis D. et al.
Assignee: Dolby Laboratories Licensing Corporation
Abstract:
A transform encoder, a transform decoder, and a transform
encoder/decoder system employ adaptive bit allocation wherein each code
word representing spectral information is allocated a fixed number of bits
and an adaptive number of bits, except that at least some but not all code
words are allocated a fixed number of bits.
Exemplary Claim [1]:
1. An encoder for the encoding of digital information, said digital
information comprising signal sample blocks representing analog audio
signals, comprising
means for generating subband information blocks, each subband
information block comprising a set of digital words generated in response
to a signal sample block, said means comprising means for applying a
discrete transform function to each of said signal sample blocks, and
means for quantizing with a fixed number of bits one or more digital
words corresponding to at least the lowest frequency spectral component of
said analog audio signals, but not quantizing all digital words with a

fixed number of bits.

3 05109417 Apr 28 1992 381/36 Low bit rate transform coder,
Dec 29 1989 decoder, and encoder/decoder for
high-quality audio

Status: certificate of correction has been issued
Inventor: Fielder; Louis D. et al.
Assignee: Dolby Laboratories Licensing Corporation
Abstract:
Quantizing noise is lessened in a speech signal system by using adaptive bit allocation for subband channels, wherein subband information of digital words is represented in block-floating-point form, and normalized mantiss as may allow dropping a sign bit.
Exemplary Claim [1]:
1. An encoder for the encoding of digital information, said digital information comprising signal sample blocks representing analog audio signals, comprising
means for defining subbands and for generating subband information blocks, each subband information block comprising a set of digital words generated in response to a signal sample block, and
means for representing one or more of said digital words in block-floating-point form comprising a mantissa associated with an exponent, for normalizing at least some mantissas, and for dropping a sign bit for a single normalized mantissa uniquely associated with an exponent whenever the value of said sign bit can be established from the value of a most significant data bit in said normalized mantissa.

4 05230038 Jul 20 1993 395/2 Low bit rate transform coder,
Nov 4 1991 decoder, and encoder/decoder for
high-quality audio

Inventor: Fielder; Louis D. et al.
Abstract:
A transform encoder, a transform decoder, and a transform encoder/decoder system utilize complex pre- and post-transform multiplication of input signal samples to implement concurrent application of a modified Discrete Cosine Transform and a modified Discrete Sine

Transform according to the Evenly-Stacked Time Domain Aliasing Cancellation technique against two channels of input signal samples, and

to reduce the computational complexity of a digital filter bank of a modified Discrete Cosine Transform.

Exemplary Claim [1]:

1. An encoder for the encoding of digital information, said digital information comprising signal sample block pairs, each of said block pairs comprising a respective one of a sequence of first signal sample blocks and a respective one of a sequence of second signal sample blocks, said sequence of first signal sample blocks representing a first channel of analog audio signals and said sequence of second signal sample blocks representing a second channel of analog audio signals, comprising

means for generating subband information block pairs in response to said signal sample block pairs by applying a function substantially corresponding to the alternate application of a modified Discrete Cosine Transform function and a modified Discrete Sine Transform function in accordance with the evenly-stacked Time-Domain Aliasing Cancellation technique, each of said subband information block pairs comprising a first subband information block and a second subband information block, said means comprising

means for multiplying real valued signal samples in the signal sample block pairs by a first complex function to obtain complex-valued modified signal sample sets,

means for applying a Fast Fourier Transform concurrently against said complex-valued modified signal sample sets, and

means for multiplying the results of the transformation by a second complex function to obtain said subband information block pairs, each first subband information block comprising a first set of digital words, and each second subband information block comprising a second set of digital words, and

means for quantizing said digital words.

5 05479562 Dec 26 1995 395/2.38 Method and apparatus for encoding
Jun 18 1993 and decoding audio information
Inventor: Fielder; Louis D. et al.
Assignee: Dolby Laboratories Licensing Corporation
Abstract:
The invention relates to formatting encoded audio information in a form suitable for transmission or storage. Audio information is encoded into a binary form, using an invariant number of bits to represent at least some but not all of the encoded information. The information represented by an invariant number of bits is assembled into pre-established positions within a formatted frame.
Exemplary Claim [1]:
1. An encoder for the encoding of audio information comprising signal samples, said encoder comprising
means for receiving said signal samples,
subband means, including adaptive bit allocation means, for generating subband information comprising digital words in response to said signal samples, wherein said adaptive bit allocation means allocates an adaptive number of bits to represent at least a portion of at least some of said digital words, and wherein at least a portion of some of said digital words is represented by an invariant number of bits, and
formatting means for assembling digital information including said digital words into a digital output having a format suitable for transmission or storage, wherein said formatting means places said portion of digital words represented by an invariant number of bits into one or more pre-established positions within a frame of said digital output.

6 05222189 Jun 22 1993 395/2 Low time-delay transform coder,
Sep 26 1990 decoder, and encoder/decoder for high-quality audio
Status: certificate of correction has been issued
Inventor: Fielder; Loius D.
Assignee: Dolby Laboratories Licensing Corporation
Abstract:
PCT No. PCT/US90/00507 Sec. 371 Date Sep. 26, 1990 Sec.

102(e) Date
Sep. 26, 1990 PCT Filed Jan. 29, 1990.

A low bit-rate (192 kBits per second) transform encoder/decoder system (44.1 kHz or 48 kHz sampling rate) for high-quality music applications employs short time-domain sample blocks (128 samples/block) so that the system signal propagation delay is short enough for real-time aural feedback to a human operator. Carefully designed pairs of analysis/synthesis windows are used to achieve sufficient transform frequency selectivity despite the use of short sample blocks. A synthesis window in the decoder has characteristics such that the product of its response and that of an analysis window in the encoder produces a composite response which sums to unity for two adjacent overlapped sample blocks. Adjacent time-domain signal samples blocks are overlapped and added to cancel the effects of the analysis and synthesis windows. A technique is provided for deriving suitable analysis/synthesis window pairs. In the encoder, a discrete transform having a function equivalent to the alternate application of a modified Discrete Cosine Transform and a modified Discrete Sine Transform according to the Time Domain Aliasing Cancellation technique or, alternatively, a Discrete Fourier Transform is used to generate frequency-domain transform coefficients. The transform coefficients are nonuniformly quantized by assigning a fixed number of bits and a variable number of bits determined adaptively based on psychoacoustic masking. A technique is described for assigning the fixed bit and adaptive bit allocations. The transmission of side information regarding adaptively allocated bits is not required. Error codes and protected data may be scattered throughout formatted frame outputs from the encoder in order to reduce sensitivity to noise bursts.

Exemplary Claim [1]:

1. An encoder for the encoding of audio information comprising signal samples, said encoder comprising

means for receiving said signal samples,
subband means, including adaptive bit allocation means, for defining subbands and for generating subband information in response to said signal samples, said subband information for each of said subbands including one or more digital words, each of said digital words comprising an adaptive portion and a non-adaptive portion, wherein coding accuracy of said adaptive portion is established by said adaptive bit allocation means, and
formatting means for assembling digital information including said subband information into a digital output having a format suitable for transmission or storage.

7 05260980 Nov 9 1993 375/241 Digital signal encoder
Aug 20 1991
Inventor: Akagiri; Kenzo et al.
Assignee: Sony Corporation
Abstract:
An apparatus for encoding an input digital signal divides the input digital signal to produce frequency components of increasing bandwidth as the frequency increases, performs blocking on each of the frequency components to produce blocks of decreasing time length as the frequency increases, performs a fast Fourier transformation on each block and quantizes each block using a quantization characteristic set in accordance with a comparison of the energy of each frequency component in each block with a masking spectrum. The masking spectrum includes both a masking effect on the frequency base of the input signal and masking effect on the time base of the input signal and may further include effects of the minimum audible curve of the human hearing sense characteristic.
Utilization of both the temporal and frequency masking effects permits reduction in the bit rate of the encoded signal using real time processing and without deterioration in sound quality.
Exemplary Claim [1]:
1. An apparatus for encoding an input digital signal,

comprising;
first frequency analysis means for frequency dividing said input digital signal into frequency components,
second frequency analysis means for blocking and frequency analyzing said frequency components to produce blocks of data having analyzed frequency components,
first noise level setting means for generating first output signals based on a first allowable noise level set in accordance with an energy of each said analyzed frequency component of said blocks of data output within a predetermined time period from said second frequency analysis means,
second noise level setting means for generating second output signals based on a second allowable noise level of each said analyzed frequency component output within said predetermined time period set in accordance with an energy of at least one of said blocks of data which temporally precede and succeed, respectively, said each block, and
quantization means for quantizing said blocks of data output from said second frequency analysis means on the basis of a quantization characteristic set in accordance with said first and second output signals from said first and second noise level setting means, respectively.

8 05241603 Aug 31 1993 381/37 Digital signal encoding apparatus
May 21 1991
Inventor: Akagiri; Kenzo et al.
Assignee: Sony Corporation
Abstract:
The present invention relates to a digital signal encoding apparatus in which an input digital signal has its frequency range divided into a number of frequency bands so that the bandwidths are broader at higher frequencies. A first allowable noise level based on the energy levels in the respective bands, and a second allowable noise level based on the energies of the signals temporally adjacent to the signals of a frequency band under consideration, are set. Signal components of each

frequency
band are quantized with a number of bits corresponding to a difference
between an output synthesized from the first and second noise levels and
the energy level of each frequency band. In this manner, the bit rate may
be lowered while deterioration in the sound quality is minimized.

Exemplary Claim [1]:

1. A digital signal encoding apparatus for encoding an input digital
signal having an input frequency range, comprising:

dividing means for dividing said input digital signal into respective
signal components within a plurality of frequency bands within the input
frequency range,

first noise level setting means for setting a first allowable noise
level for the respective signal components within each of said plurality
of frequency bands based on energy levels of respective signal components
occurring concurrently therewith and within at least one frequency band
other than said each of said plurality of frequency bands,

second noise level setting means for setting a second allowable noise
level for the respective signal components within said each of said
plurality of frequency bands based on energy levels of temporally adjacent
signals in the same band as said respective signal components,

synthesizing means for generating a synthesized allowable noise level
for the respective signal components within said each of said plurality of
frequency bands based on the first and second allowable noise levels set
therefor by said first noise level setting means and said second noise
level setting means, respectively, and

quantizing means for quantizing the respective signal components of
said each of said plurality of frequency bands with a bit number based on
said synthesized allowable noise level generated therefor.

9 05414795 May 9 1995 395/2.13 High efficiency digital data
Mar 26 1992 encoding and decoding apparatus
Inventor: Tsutsui; Kyoya et al.

Assignee: Sony Corporation
Abstract:
An apparatus for compressing a digital input signal compresses a digital input signal arranged into frames of plural samples. The digital input signal is orthogonally transformed by plural orthogonal transform circuits in blocks derived by dividing the frames by a different divisor in each circuit. The resulting spectral coefficients are quantized using an adaptive number of bits. The output of one orthogonal transform circuit is selected based on the outputs of the orthogonal transform circuits. The digital input signal is divided into plural frequency ranges, and frames are formed in each range. A block length decision circuit determines division of the frames of each range signal into blocks in response to range signal dynamics. The range signals are orthogonally transformed in blocks and the spectral coefficients are quantized. The apparatus also includes a frequency analyzing circuit that derives spectral data points from the input data, and groups them in plural bands. A noise level setting circuit finds the energy in each band and sets a allowable noise level in response to the band energy. A block floating coefficient calculating circuit calculates block floating coefficients based on the maximum spectral data point in each band. A quantizing bit number decision circuit determines the number of bits to use for quantizing in response to the allowable noise level and the block floating coefficients. A quantizing circuit quantizes the spectral data points in response to the quantizing bit number decision circuit.
Exemplary Claim [3]:
3. An apparatus for compressing a digital input signal according to claim 1, wherein:
each orthogonal transform means quantizes the spectral coefficients using a primary number of bits and provides as the selection signal information indicating the primary number of bits; and

the selection means selects as the compressed signal the output signal
from the orthogonal transform means providing the selection signal
indicating the least primary number of bits.

US006339757B1

(12) **United States Patent**
Teh et al.

(10) **Patent No.: US 6,339,757 B1**
(45) **Date of Patent: Jan. 15, 2002**

(54) **BIT ALLOCATION METHOD FOR DIGITAL AUDIO SIGNALS**

(75) Inventors: **Do Hui Teh; Pek Yew Tan; Sua Hong Neo**, all of Singapore (SG)

(73) Assignee: **Matsushita Electric Industrial Co., Ltd.**, Osaka (JP)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

(21) Appl. No.: **08/161,798**

(22) Filed: **Dec. 6, 1993**

(30) **Foreign Application Priority Data**

Feb. 19, 1993 (JP) 5-030712

(51) **Int. Cl.**[7] **G10L 19/04**
(52) **U.S. Cl.** **704/219**; 704/229; 704/220
(58) **Field of Search** 395/2.38, 2.39; 381/29, 30

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WO	WO92/17884	10/1992

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IEEE Global Telecommunications Conference, vol. 1, Dec. 2, 1990, New York, pp. 518–522, Mahieux, et al. "Transform Coding of Audio Signals at 64KBIT/S[1]," p. 520, ¶V "The Bit Allocation."

Philips Journal of Research, vol. 44, No. 2/3, Jul. 1989, Eindhoven NL, pp. 329–343, Veldhuis, et al. "Subband Coding of Digital Audio Signals," pp. 338, 1.3—p. 340, 1.33.

* cited by examiner

Primary Examiner—Fan Tsang
Assistant Examiner—Michael N. Opsasnick
(74) *Attorney, Agent, or Firm*—Smith, Gambrell & Russell, LLP

(57) **ABSTRACT**

When determining a quantization for digital audio signals having a spectral and temporal structure wherein the audio signals are buffered in frames and decomposed into spectral components, the method is: obtaining a variance and/or a representative of the components of the the frame within a defined frequency interval; computing a necessary bandwidth using psychoacoustic criteria; determining an initial quantization for each frequency interval using an approximate mathematical model; and iteratively increasing the quantization of each of the frequency intervals through a psychoacoustically weighted inverse relationship with the the variance of each of the frequency intervals.

21 Claims, 8 Drawing Sheets

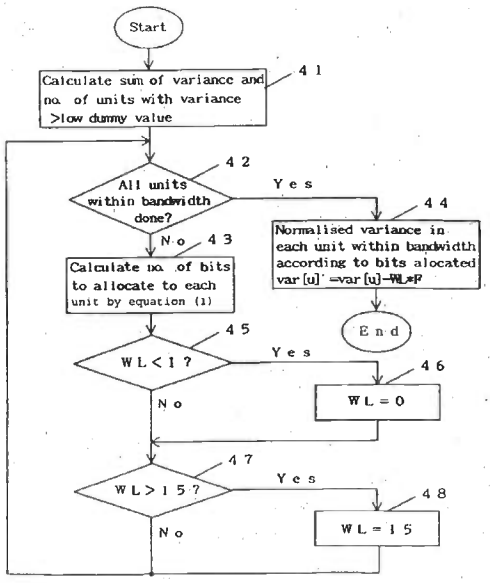


Fig. 1

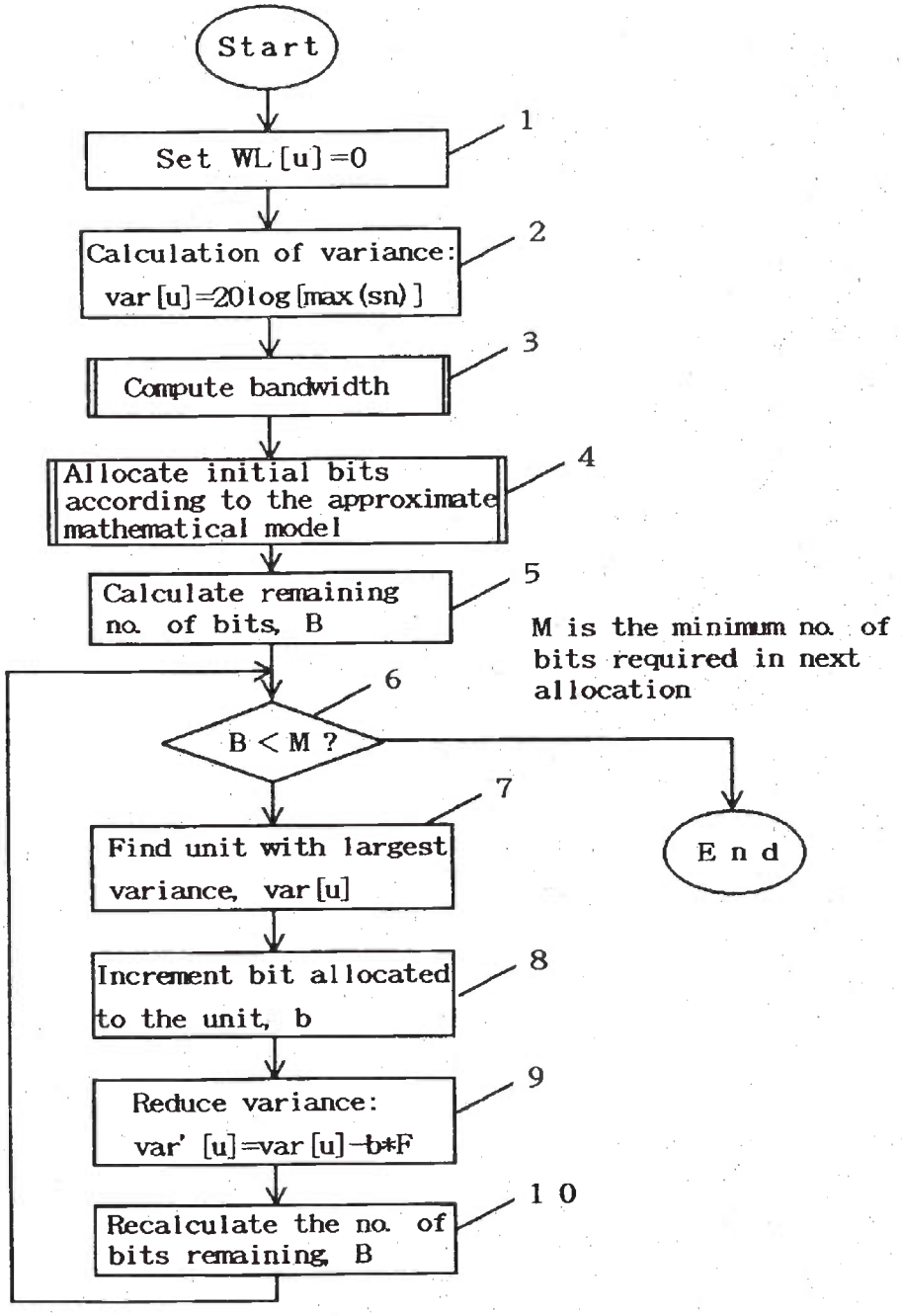

Start
Set WL[u]=0
1
Calculation of variance:
var[u]=20log[max(sn)]
2
Compute bandwidth
3
Allocate initial bits according to the approximate mathematical model
4
Calculate remaining no. of bits, B
5
M is the minimum no. of bits required in next allocation
6
B < M ?
End
7
Find unit with largest variance, var[u]
Increment bit allocated to the unit, b
8
Reduce variance:
var' [u]=var[u]-b*F
9
Recalculate the no. of bits remaining, B
10

Fig. 2

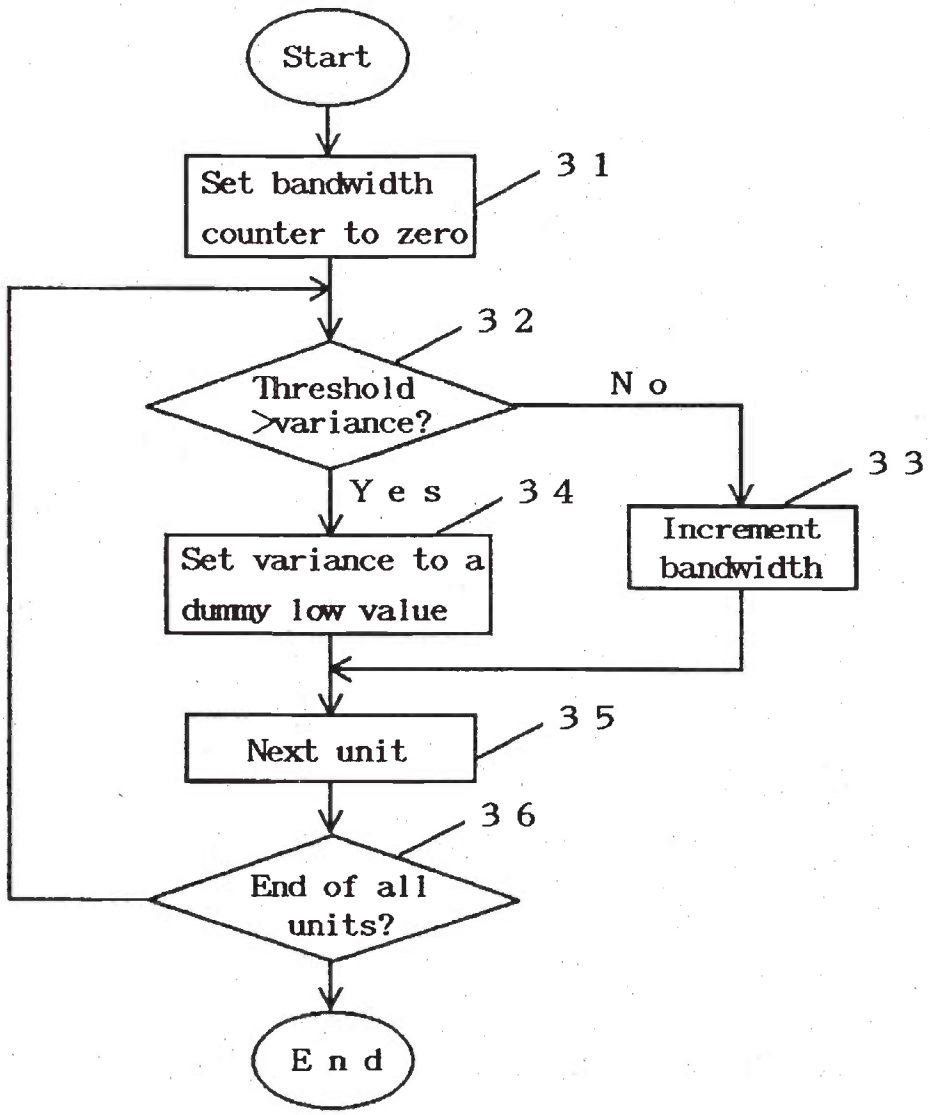
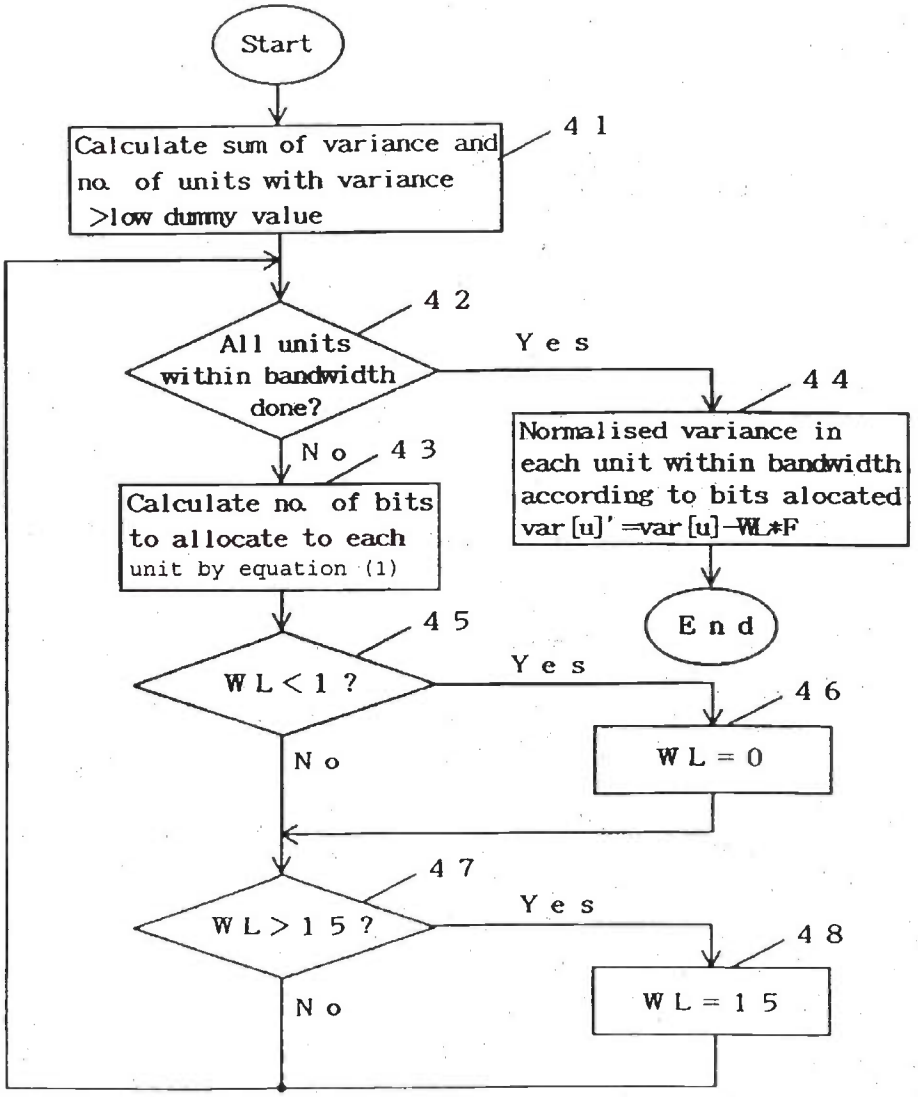

Start
Set bandwidth counter to zero
31
32
Threshold >variance?
No
Yes
33
34
Increment bandwidth
Set variance to a dummy low value
Next unit
35
36
End of all units?
End

Fig. 3


Start
Calculate sum of variance and no. of units with variance >low dummy value
41
All units within bandwidth done?
42
Yes
No
43
Calculate no. of bits to allocate to each unit by equation (1)
44
Normalised variance in each unit within bandwidth according to bits alocated var[u]' =var[u]-WL*F
End
WL<1?
45
Yes
No
46
WL=0
WL>15?
47
Yes
No
48
WL=15

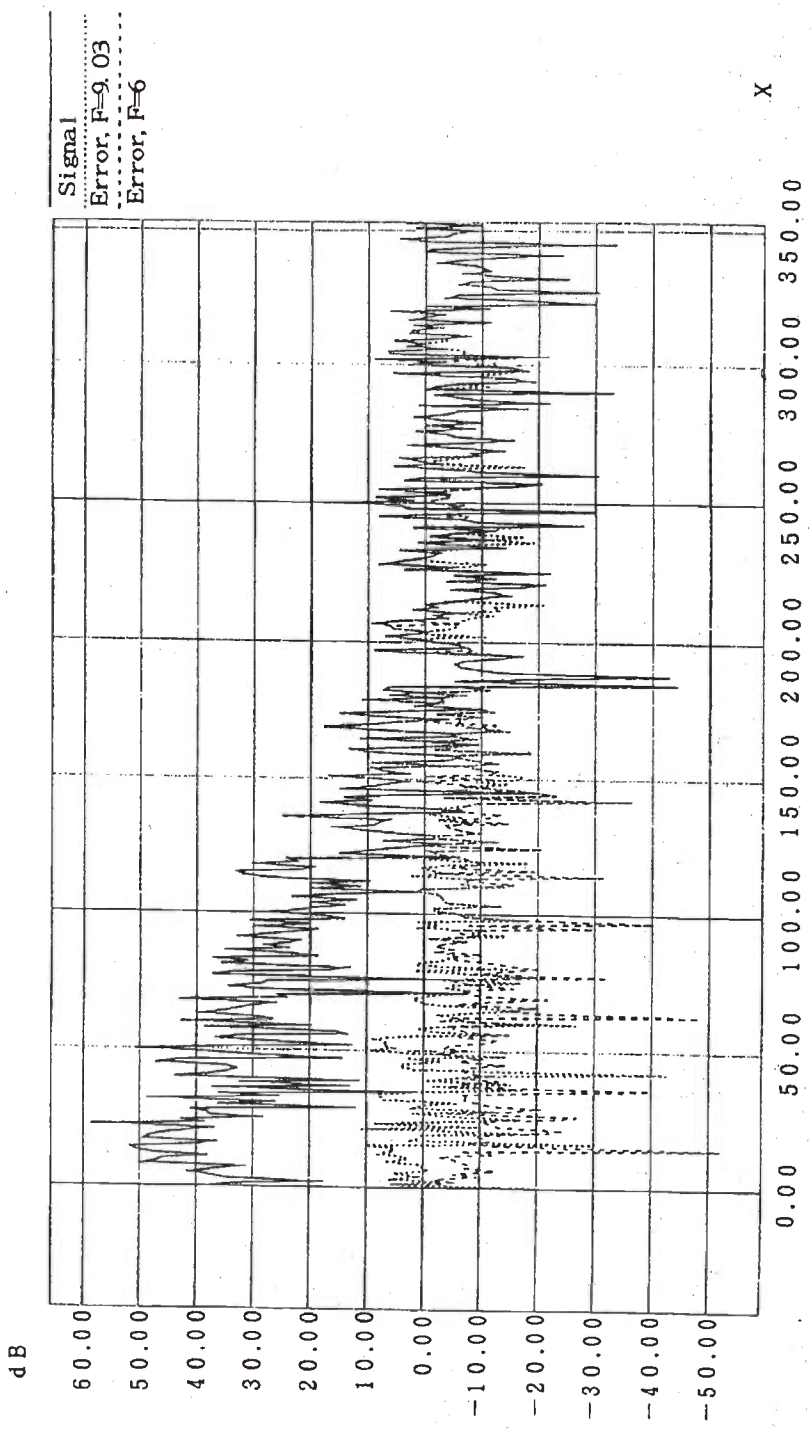

Signal
Error, F=9.03
Error, F=6
X
dB
60.00
50.00
40.00
30.00
20.00
10.00
0.00
-10.00
-20.00
-30.00
-40.00
-50.00
0.00
50.00
100.00
150.00
200.00
250.00
300.00
350.00

Fig. 4

Fig. 5

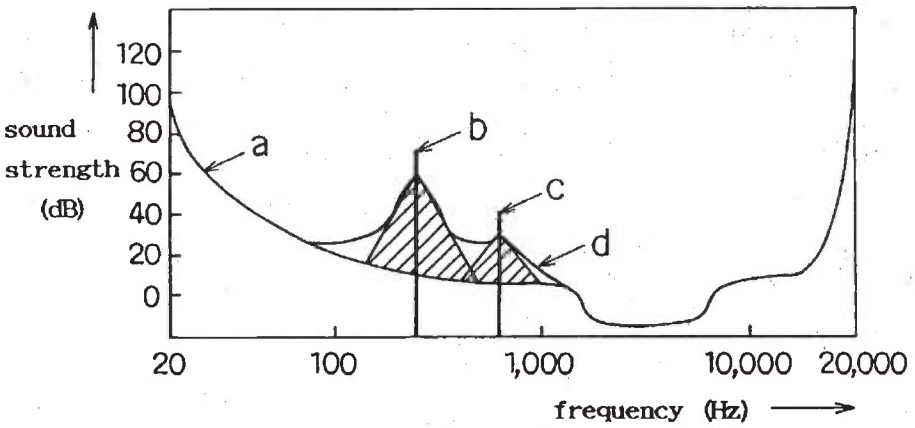

sound strength (dB)
120
100
80
60
40
20
0
a
b
c
d
20
100
1,000
10,000
20,000
frequency (Hz)

Fig. 6

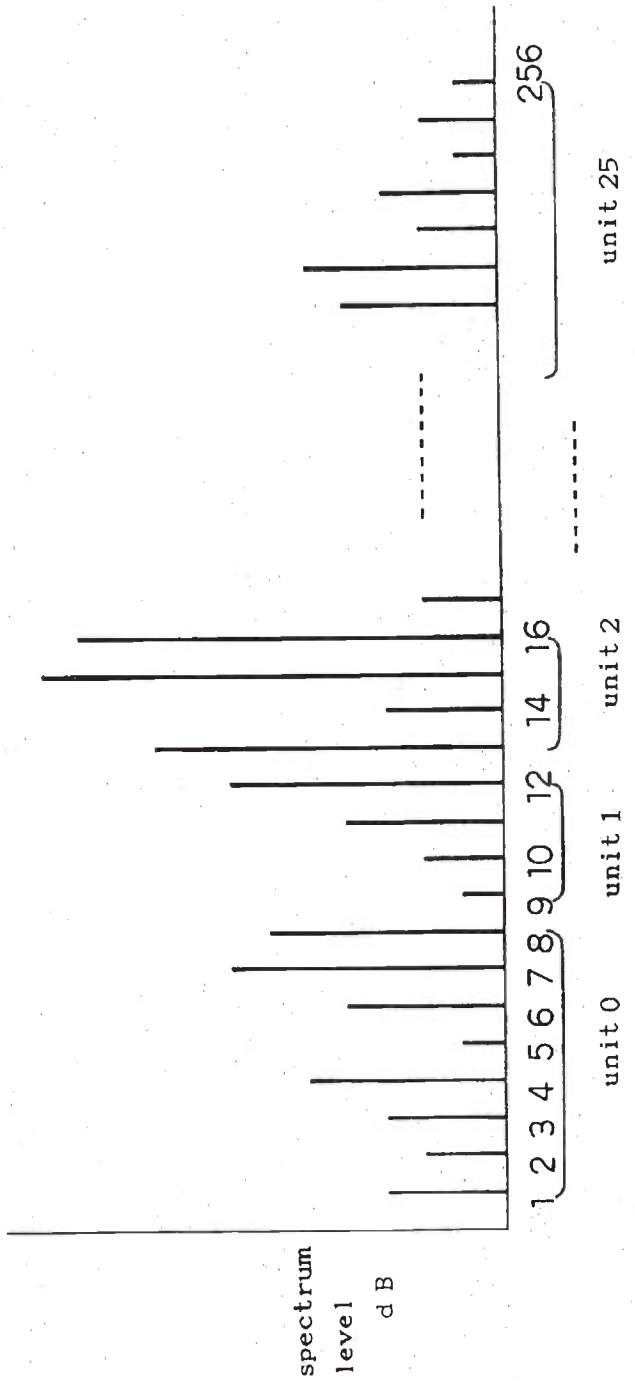

spectrum level
dB
1 2 3 4 5 6 7 8 9 10 12 14 16
256
unit 0
unit 1
unit 2
unit 25

Fig. 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)

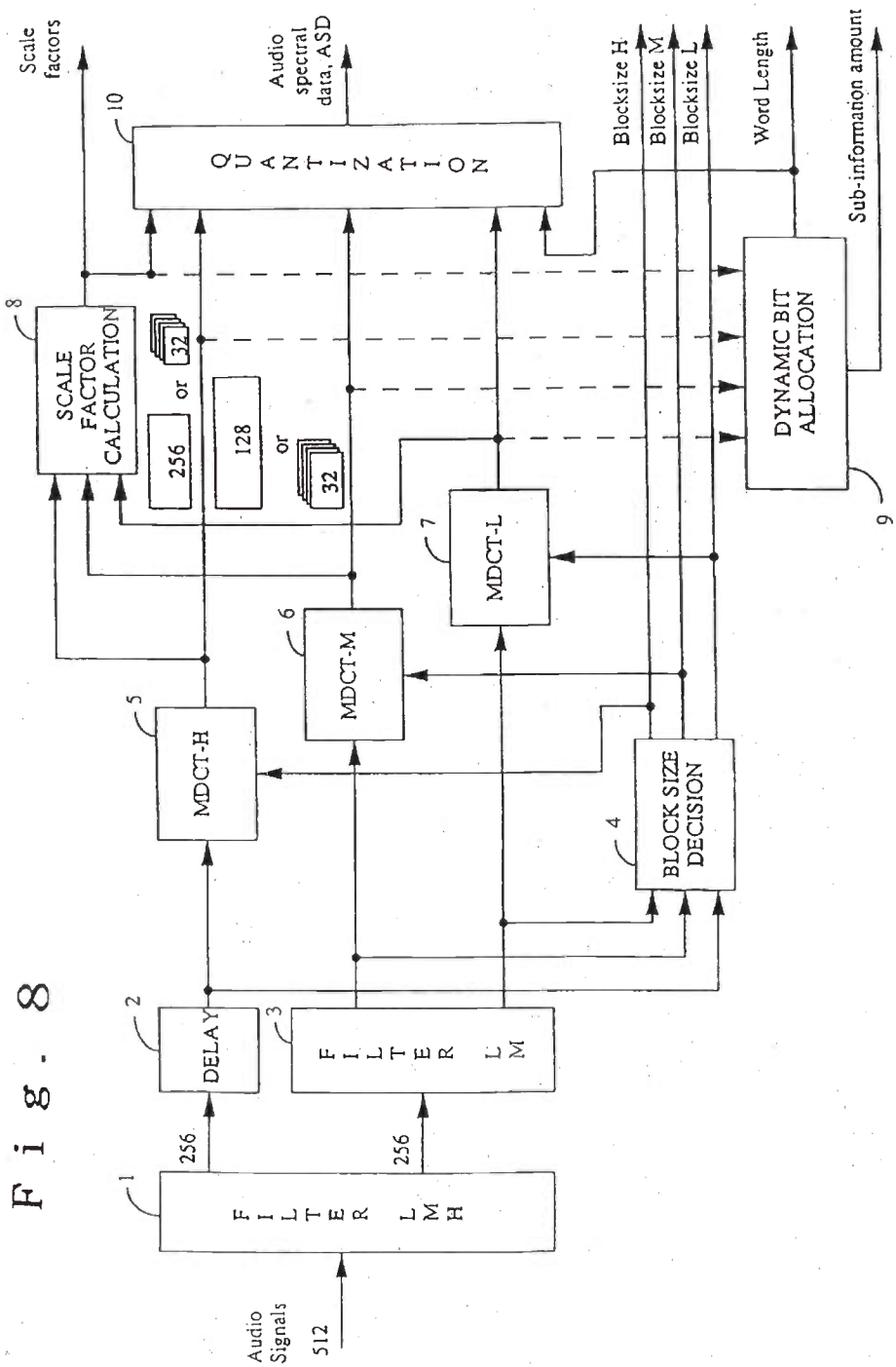

Audio Signals
512
FILTER
L M H
1
256
256
DELAY
2
FILTER
L M
3
BLOCK SIZE DECISION
4
MDCT-H
5
MDCT-M
6
MDCT-L
7
SCALE FACTOR CALCULATION
8
DYNAMIC BIT ALLOCATION
9
QUANTIZATION
10
Scale factors
Audio spectral data, ASD
Blocksize H
Blocksize M
Blocksize L
Word Length
Sub-information amount
256
or
32
128
or
32

Fig. 8

BIT ALLOCATION METHOD FOR DIGITAL AUDIO SIGNALS

BACKGROUND OF THE INVENTION

1. Field of The Invention

This invention relates to the efficient information coding of digital audio signals for transmission or digital storage media especially a quantization method using an efficient bit allocation method for information coding

2. Related art of The Invention

Compression algorithms for wide band audio signals are finding wide-spread applications in recent years. In the area of consumer electronics, the digital compact cassette (DCC) and the mini-disc system (MD) are two applications of these compression algorithms.

While the DCC system uses a compression algorithm that is based on sub-band coding, the MD system uses an algorithm that is based on a hybrid of sub-band and transform coding with the transform coding portion forming the backbone of the algorithm. This invention is related to the dynamic bit allocation of the MD coder.

The MD system uses the ATRAC compression algorithm that is documented in chapter 10 of the MD system description by Sony in December 1991. The ATRAC algorithm compresses the input audio signals at a bit rate of 705.6 k bit/s/channel to a bit rate of 146.08 k bit/s/channel.

FIG. **8** shows the block diagram of the encoding process. The input time signals are first passed through a splitting filter bank, **1**, **2**, **3**, to obtain the signals in three frequency bands. The lower two bands are each at half the bandwidth of the uppermost band. Block size decision, **4**, is made for each band to determine the sample size or block mode for the windowing and transform process, **5**, **6**, **7**. One of the two block modes available—short block mode or long block mode, will be selected for each of the bands. The 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. Quantization, **10**, is carried out on the samples using the scale factor and bit allocation information from the dynamic bit allocation module, **9**.

The dynamic bit allocation method forms an integral part in any adaptive compression algorithm. The quality of the reconstructed output and the extent of redundancy and irrelevancy removal are largely determined by the bit allocation method. In addition, the bit allocation procedure also plays a part in determining the degree of hardware complexity. As the bit allocation in the ATRAC algorithm is mainly applied to the transformed spectral samples, numerous example of prier art exist for this type of transform coder. These dynamic bit allocation techniques can be grouped largely into two categories.

The first category consists of bit allocation methods applying the psychoacoustic phenomena of simultaneous masking and threshold in quiet to derive the masking threshold. Examples in this category include the bit allocation in the MD system description, where the MDCT spectral samples are used to compute the masking threshold. A more complicated technique is described in the paper entitled 'Transform Coding of Audio Signals using perceptual noise criteria' by J. D. Johnston, where a Fast Fourier Transform (FFT) is used to obtain the frequency spectral components for more complex masking calculations.

The second category offers a more simplified method of allocating according to the signal statistics. An example is the optimum bit allocation procedure described in 'Digital Coding of Waveforms' by N. S. Jayant and P. No **11**, which allocates the bits to the different spectral components by minimizing the reconstructed error against a constant bit rate.

Meanwhile the object of the dynamic bit allocation procedure is to remove redundancy and irrelevancy while maintaining the original audio quality. To enable the applications of this algorithm in consumer products, which in general are low in cost, the algorithm has to have low complexity. The 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. The bit allocation procedures described under category one in the previous section are able to produce good sound quality. However, a great deal of complexity is incurred. Most of these algorithms require full DSP (either fixed point or floating point in some cases) power in order to perform the bit allocation.

On the other hand, the optimum bit allocation procedure mentioned above, while less complex, is unable to attain the quality of the perceptual based bit allocation as its design is concentrated purely on the source which is the audio signal and not on the final receiver, the human auditory system.

Further, allocation procedure requires a large number of iterative loops in order to use up the channel bit rate optimally. This problem was highlighted by the authors in the 'Digital Coding of Waveforms'. Therefore this bit allocation algorithm is unsuitable in the case where there is a great constraint on the execution time of the DSP. To compound the problem of intensive iteration is the fact that in the MD coding algorithm, a non-uniform number of spectral samples are grouped in each unit, therefore making it difficult to compute the number of bits that will be used up without a bit by bit allocation.

SUMMARY OF THE INVENTION

The object of this invention is to design a dynamic bit allocation procedure that is adapted to the human auditory system and yet with a low level of complexity so that high audio quality can be achieved while at the same time meeting the low cost target.

For the purpose of achieving the above object, the dynamic bit allocation comprises the means of obtaining the variance or a representative within a defined frequency interval as an accurate representation of the signals in the interval; the means of determining the necessary bandwidth of the audio signal using the human hearing threshold so that irrelevancy is removed; the means of determining the initial quantizations using an approximate mathematical model which considers the in-band masking effect of the signal; the means of increasing the quantizations iteratively so as to meet the final bit rate required while maintaining the affect of the mathematical model.

The bit allocation method has a very important role for determining the quality of the reconstructed output and the complexity of hardware. The present invention realizes a dynamic bit allocation method which produces low hardware complexity and high quality audio, by combining psychoacoustic criteria and human hearing threshold. The low complexity hardware is realized by allocating an initial bit number using the approximations made in the mathematical model before the remaining bit number is allocated by a precise iterative process.

That is in the present invention, the variance or representative used in the bit allocation procedure considers the dynamic behavior of the signals and adjusts to it while the bandwidth computation considers the human auditory system and adapts the bit allocation to it. The mathematical model also adapts to the human auditory system by considering masking while at the same time serving to reduce the number of computational steps or loops required for the bit allocation. This means further simplifying the procedure by approximating the dynamic elements within the relation with the statistically derived constants. The iterative means ensure that the desired bit rate is met while adjusting for the approximations made in the mathematical model.

Meanwhile the meaning of masking threshold is described as follows. FIG. **5** shows an example of the masking threshold. In the figure 'a' indicates the least audible value and the area above the line is the human audible area. And 'b' and 'c' shows spectrum of strong signal component and the hatched parts are the such portion masked by signals 'b' and 'c'. The curve 'd' is a threshold value of masking obtained from the audible area and the hatched parts masked by 'b', 'c' signals. Since the signal below the threshold value 'd' can not be sensed, the quantization is not necessary. Since the masking threshold is high at signal 'b', a higher quantization noise near this frequency will still remain inaudible.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. **1** is the flowchart of the present embodiment of the invention.

FIG. **2** is the flowchart of the compute bandwidth module within the present embodiment.

FIG. **3** is the flowchart of the initial bit allocation module within the present embodiment.

FIG. **4** is the spectral plot of a frame of violin signals and the error spectrums obtained with different values of F.

FIG. **5** is the graph showing the relation of the frequency and sound strength (dB).

FIG. **6** is the illustration of a of spectrum grouped into 26 units.

FIG. **7** is the table of the bit allocation data WL.

FIG. **8** is the block diagram of the encoding process of the prior art.

PREFERRED EMBODIMENTS

FIG. **1** shows an embodiment of the present invention. First, the quantizations or bit allocations for each frequency interval or unit (WL[u]) is set to zero (step numeral **1**). The details of the unit are described in the next paragraph. The variance in each unit is calculated from the maximum spectral sample amplitude in the unit (step of numeral **2**). Other embodiments the invention can have variations in step numeral **2** where the variance can be calculated from an average of the spectral components within the unit. The maximum spectral sample amplitude was found to be a good worst case representative of the signals in the unit, be it tone-like or noise-like. Further, using the maximum spectral sample saves some computational steps. The bandwidth of transmission is computed next (step numeral **3**). In step numeral **4**, the initial number of bits are allocated according to the approximate mathematical model. Calculation of the remaining bits is then done in step numeral **5**. The remaining bits, B are iteratively allocated to the units according to the following steps until the number of bits remaining is less than the minimum number of bits required in the next allocation.

[1] If the remaining bits, B is less than the minimum number of bits required in the next allocation, then the iteration is stopped (step numeral **6**).

[2] Find the unit with the largest variance, var (step numeral **7**).

[3] Increment the bit allocations in the unit by one or two bits (b),

WL[u]=WL[u]+b (step numeral **8**).

[4] Reduce the variance of the unit by var[u]=var[u]−b*F, where F is the psychoacoustic weighting factor(step numeral **9**).

[5] Recalculate the number of bits remaining, B'=B−b*A[u], where A[u] is the number of spectral lines in the unit, (step numeral **10**).

[6] Steps [6] to [10]are repeated.

The unit, spectrum number and bit allocation data (WL) are described as follows.

The audio signal is transformed into, for example, 256 spectral lines by the MDCT. The spectral lines are grouped into plural units. In FIG. **6**, the 256 spectral lines are grouped into **26** number units from 0–25. The bit allocation data WL(u) is given by an unit as shown in FIG. **7**. In the example of FIGS. **6**, **7**, the spectral line numbers **1**–**8** have the same bit allocation data WL(**0**) and the spectral line numbers **9**–**12** have same bit allocation data WL(**1**).

FIG. **2** shows the flow chart of the bandwidth computation procedure, a modified set of values of the human hearing threshold, obtained from listening tests, is used to determine the bandwidth of the audio signal and to almost eliminate any units where the variance is below the threshold from the next stage of bit allocation. First, the bandwidth counter is set to zero (step numeral **31**). In step numeral **32**, the block mode is tested. If the short block mode was used in the MDCT process, the thresholds for units having the same frequency resolution is obtained by selecting the lowest threshold from all these units (step numeral **33**). Each unit's variance is then compared against the threshold (step numeral **34**). If the variance is above the threshold, the bandwidth counter is incremented (step numeral **35**). Otherwise, the variance in the unit is set to a low dummy value so that it will have the least chance of being allocated any bits in the next stage (step numeral **36**). This procedure is repeated for all units (step of numeral **37** and **38**). The use the hearing threshold will eliminate units that are irrelevant according to psychoacoustic criteria.

FIG. **3** shows the procedure of allocating the initial bits to each unit before the iterative process by an approximate mathematical model. First, the sum of all the variances and the number of units with variances greater than the low dummy value are calculated (step numeral **41**). Step numeral **42** checks if all the units within the bandwidth have been processed. If all units are processed, then the variance in each unit is normalized according to the number of bits allocated to the unit (var[u]=var[u]−WL*F) (step numeral **44**). The procedure is then ended. If all of the units are not processed, then the number of bits to be allocated to the next unit, WL[u] is calculated by equation (1) shown below (step numeral **43**):

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

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

Steps numerals **45, 46, 47** and **48** check if the bit allocations are below the lowest limit or exceed the highest limit. In either the case, the lowest limit (0 bits) and the highest limit (15 bits) one is set respectively.

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. However, in the higher frequency end, the error spectrum of F=9.03 is lower and this accounts for the improvement in the noise control at high frequency. This form of allocation considers that strong signals can have greater in-band masking than weak signals and it is more advantageous to vary the error spectrum more closely to the signal spectrum. The mathematical model is able to allocate up to at least 65% of the total number of bits, thus reducing the number of iterations required.

This invention has been simulated to produce very high audio quality when used in the ATRAC algorithm. The bit allocation technique, being computationally less intensive takes up a total of 5.7 Mips. This can contribute significantly to a low cost and low power (due to the low frequency of operation) single chip LSI implementation of the encoder.

What is claimed is:

1. 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:

a step for dividing at least one frequency interval into a plurality of frequency-units;

a step for obtaining a representative of each frequency-unit obtained;

a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria;

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

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

2. Method according to claim **1**, wherein

said mathematical model is derived from an in-band masking effect such that an error spectrum will follow more closely to the original signal spectrum.

3. Method according to claim **2**, wherein each frequency-unit is represented by a single spectral component.

4. Method according to claim **2**, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.

5. Method according to claim **2**, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.

6. A method according to claim **1**, wherein said second bit allocation step is to determine the bit allocation through a psychoacoustically weighted inverse relationship with the said representative of each of said frequency-units.

7. Method according to claim **6**, wherein each frequency-unit is represented by a single spectral component.

8. Method according to claim **6**, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.

9. Method according to claim **6**, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.

10. A method according to claim **1**, wherein said counted number of available frequency-units is a number of the frequency-units which have the representative exceeding a human hearing threshold.

11. Method according to claim **10**, wherein each frequency-unit is represented by a single spectral component.

12. Method according to claim **10**, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.

13. Method according to claim **10**, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.

14. Method according to claim **1**, wherein each frequency-unit is represented by a single spectral component.

15. Method according to claim **1**, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.

16. Method according to claim **1**, wherein each frequency-unit is obtained by dividing said spectral and temporal structure with at least one frequency interval.

17. Method according to claim **1**, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.

18. 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:

a step for dividing at least one frequency interval into a plurality of frequency-units;

a step for obtaining a representative of each frequency-unit obtained;

a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria where the representative exceeds a human hearing threshold;

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

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

wherein said mathematical model is represented by a formula:

$$WL(u) = \frac{\mathrm{var}(u)}{F} - \frac{1}{N \cdot F} \sum_{u=0}^{N} \mathrm{var}(u) + \frac{Ba}{A \cdot N}$$

wherein

WL(u) represents number of bits allocated to the frequency-units u,

var(u) represents variance of the frequency-units u,

F represents a psychoacoustic weighting factor,

N represents the counter number of the available frequency-units,

Ba represents number of bits set aside for initial allocation, and

A represents approximate average number of spectral samples in the available frequency-units.

19. Method according to claim **18**, wherein each frequency-unit is represented by a single spectral component.

20. Method according to claim **18**, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.

21. Method according to claim **18**, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.

* * * * *

PATENT APPLICATION SERIAL NO. 08/161798

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PATENT AND TRADEMARK OFFICE
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BEVERIDGE, DEGRANDI, WEILACHER & YOUNG
ATTORNEYS AND COUNSELORS AT LAW
SUITE 800
1850 M STREET, N. W.
WASHINGTON, D. C. 20036
(202) 659-2811
FACSIMILE: (202) 659-1462
TELEX: WUI 64470

JOSEPH A. DeGRANDI
ROBERT G. WEILACHER
RICHARD G. YOUNG
JOHN T. ROBERTS
MICHAEL A. MAKUCH
BERNARD A. MEISLING
HELEN M. McCARTHY
LEROY G. SINN
HELEN H. MINSKER *
DENNIS C. RODGERS
G. BYRON STOVER *
WILLIAM F. RAUCHHOLZ
MAURICE U. CAHN [5]
*NOT ADMITTED IN D.C.

ANDREW B. BEVERIDGE
1916-1972

OF COUNSEL
YOICHIRO YAMAGUCHI *
DRAGOSLAV RISTIC *
*JAPAN ONLY
*YUGOSLAVIA ONLY

PATENT AGENTS
THOMAS L. EVANS
QIXIA ZHANG *
ATTILA MANDI *
*HUNGARY ONLY
*CHINA ONLY

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December 6, 1993

Hon. Commissioner of Patents and Trademarks
Washington, D.C. 20231

PATENT APPLICATION TRANSMITTAL LETTER

Inventors: **Do Hui TEH, Pek Yew TAN and Sua Hong NEO**
Title: **A BIT ALLOCATION METHOD FOR TRANSFORM CODER**
Attorney Docket No.: 03-48683

Sir:

Transmitted herewith for filing are the following: (1) new patent application including 16 pages of text, a signed Declaration, and 8 sheets of formal drawings on plain paper; (2) an Assignment Transmittal Letter and Assignment for recordation, to Mastsushita Electric Industrial Co., Ltd.; (2) a Claim for Priority with a certified copy of Application No. 5-030,712, filed in Japan on February 19, 1993; and (3) Counsel's check for the fee which has been calculated as shown below:

Basic Fee		$710.00
Total Claims	(4 less 20 = 0) x ($22)	0
Independent Claims	(1 less 3 = 0) x ($74)	0
Multiple Dependent Claim charge ($230)		0
	SUBTOTAL	$710.00
Assignment recordation fee ($40.00)		40.00
	TOTAL	$750.00

If any additional fees associated with this communication are required, please notify the undersigned attorney by telephone and charge the fees to Deposit Account 02-4300. This includes, for example, any additional filing fees required under 37 C.F.R. § 1.16 and any patent application processing fees under 37 C.F.R. § 1.17.

Respectfully submitted,

Joseph A. DeGrandi

Joseph A. DeGrandi, Reg. No. 17,446

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08/161738

TITLE OF THE INVENTION

A bit allocation method for transform coder

BACKGROUND OF THE INVENTION

1. Field of The Invention

This invention relates to the efficient information coding of digital audio signals for transmission or digital storage media , especially a quantization method using an efficient bit allocation method for information coding .

2. Related art of The Invention

Compression algorithms for wide band audio signals are finding wide-spread applications in recent years. In the area of consumer electronics, the digital compact cassette (DCC) and the mini-disc system (MD) are two applications of these compression algorithms.

While the DCC system uses a compression algorithm that is based on sub-band coding, the MD system uses an algorithm that is based on a hybrid of sub-band and transform coding with the transform coding portion forming the backbone of the algorithm. This invention is related to the dynamic bit allocation of the MD coder.

The MD system uses the ATRAC compression algorithm that is

documented in chapter 10 of the MD system description by Sony in December 1991. The ATRAC algorithm compresses the input audio signals at a bit rate of 705.6 k bit/s/channel to a bit rate of 146.08 k bit/s/channel.

Fig. 8 shows the block diagram of the encoding process. The input time signals are first passed through a splitting filter bank, 1, 2, 3, to obtain the signals in three frequency bands. The lower two bands are each at half the bandwidth of the uppermost band. Block size decision, 4, is made for each band to determine the sample size or block mode for the windowing and transform process, 5, 6, 7. One of the two block modes available - short block mode or long block mode, will be selected for each of the bands. The 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. Quantization, 10, is carried out on the samples using the scale factor and bit allocation information from the dynamic bit allocation module, 9.

The dynamic bit allocation method forms an integral part in any adaptive compression algorithm. The quality of the reconstructed output and the extent of redundancy and irrelevancy removal are

largely determined by the bit allocation method. In addition, the bit allocation procedure also plays a part in determining the degree of hardware complexity. As the bit allocation in the ATRAC algorithm is mainly applied to the transformed spectral samples, numerous examples of prior art exist for this type of transform coder. ~~quite a number~~ These dynamic bit allocation techniques can be grouped largely into two categories.

The first category consists of bit allocation methods applying the psychoacoustic phenomena of simultaneous masking and threshold in quiet to derive the masking threshold. Examples in this category include the bit allocation in the MD system description, where the MDCT spectral samples are used to compute the masking threshold. A more complicated technique is described in the paper entitled 'Transform Coding of Audio Signals using perceptual noise criteria' by J. D. Johnston, where a Fast Fourier Transform (FFT) is used to obtain the frequency spectral components for more complex masking calculations.

The second category offers a more simplified method of allocating according to the signal statistics. An example is the optimum bit allocation procedure described in 'Digital Coding of Waveforms' by N. S. Jayant and P. Noll, which allocates the bits to the different spectral components by minimizing the reconstructed error against a constant bit rate.

Meanwhile the object of the dynamic bit allocation procedure is to remove redundancy and irrelevancy while maintaining the original audio quality. To enable the applications of this algorithm in consumer products, which in general are low in cost, the algorithm has to have low complexity. The 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. The bit allocation procedures described under category one in the previous section are able to produce good sound quality. However, a great deal of complexity is incurred. Most of these algorithms require ~~a~~ full DSP~~s~~ (either fixed point or floating point in some cases) ~~processing~~ power in order to perform the bit allocation.

On the other hand, the optimum bit allocation procedure mentioned above, while less complex, is unable to attain the quality of the perceptual based bit allocation as its design is concentrated purely on the source which is the audio signal and not on the final receiver, the human auditory system.

Further, allocation procedure requires a large number of iterative loops in order to use up the channel bit rate optimally. This problem was highlighted by the authors in the 'Digital Coding of Waveforms'. Therefore ~~his~~ this bit allocation algorithm is unsuitable in the case where there is a great constraint on the execution time of the DSP.

To compound the problem of intensive iteration is the fact that in the MD coding algorithm, a non-uniform number of spectral samples are grouped in each unit, therefore making it difficult to compute the number of bits that will be used up without a bit by bit allocation.

SUMMARY OF THE INVENTION

The object of this invention is to design a dynamic bit allocation procedure that is adapted to the human auditory system and yet with a low level of complexity so that high audio quality can be achieved while at the same time meeting the low cost target.

For the purpose of achieving the above object, the dynamic bit allocation comprises the means of obtaining the variance or a representative within a defined frequency interval as an accurate representation of the signals in the interval; the means of determining the necessary bandwidth of the audio signal using the human hearing threshold so that irrelevancy is removed; the means of determining the initial quantizations using an approximate mathematical model which considers the in-band masking effect of the signal; the means of increasing the quantizations iteratively so as to meet the final bit rate required while maintaining the ~~effect~~ affect of the mathematical model.

The bit allocation method has a very important role for

determining the quality of the reconstructed output and the complexity of hardware. The present invention realizes a dynamic bit allocation method which produces low hardware complexity and high quality audio , by combining psychoacoustic criteria and human hearing threshold . The low complexity hardware is realized by allocating an initial bit number using the approximations made in the mathematical model before the remaining bit number is allocated by a precise ~~interative~~ iterative process.

That is in the present invention , the variance or representative used in the bit allocation procedure considers the dynamic behavior of the signals and adjusts to it while the bandwidth computation considers the human auditory system and adapts the bit allocation to it. The mathematical model also adapts to the human auditory system by considering masking while at the same time serving to reduce the number of computational steps or loops required for the bit allocation. This means further ~~simplifies~~ simplifying the procedure by approximating the dynamic elements within the relation with the statistically derived constants. The iterative means ensure that the desired bit rate is met while adjusting for the approximations made in the mathematical model.

Meanwhile the meaning of masking threshold is described as follows. Fig.5 shows an example of the masking threshold. In the figure 'a' indicates the least audible value and the area above the line is the human

audible area . And 'b' and 'c' shows spectrum of strong signal component and the hatched parts are ~~such~~ the portion masked by signals 'b' and 'c'. The curve 'd' is a threshold value of masking obtained from the audible area and the hatched parts masked by 'b','c' signals. Since the signal below the threshold value 'd ' can not be sensed , the quantization is not necessary. Since the masking threshold is high at signal 'b', a higher quantization noise near this frequency will still remain inaudible.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is the ~~flow-chart~~ flowchart of the present embodiment of the invention.

Fig. 2 is the ~~flow-chart~~ flowchart of the compute bandwidth module within the present embodiment.

Fig. 3 is the ~~flow chart~~ flowchart of the initial bit allocation module within the present embodiment.

Fig. 4 is the spectral plot of a frame of violin signals and the error spectrums obtained with different values of F.

Fig. 5 is the graph showing the relation of the frequency and sound strength ~~(DB)~~ (dB).

Fig. 6 is the illustration of a spectrum grouped into 26 units.

Fig. 7 is the table of the bit allocation data WL.

Fig. 8 is the block diagram of the encoding process of the prior art.

PREFERRED EMBODIMENTS

Fig. 1 shows an embodiment of the present invention. First, the quantizations or bit allocations for each frequency interval or unit (WL[u]) is set to zero (step of numeral 1). The details of the unit are described in the next paragraph. The variance in each unit is calculated from the maximum spectral sample amplitude in the unit (step of numeral 2). Other embodiments of the invention can have variations in step of numeral 2 where the variance can be calculated from an average of the spectral components within the unit. The maximum spectral sample amplitude was found to be a good worst case representative of the signals in the unit, be it tone-like or noise-like. Further, using the maximum spectral sample saves some computational steps. The bandwidth of transmission is computed next (step of numeral 3). In step of numeral 4, the initial number of bits are allocated according to the approximate mathematical model. Calculation of the remaining bits is then done in step of numeral 5. The remaining bits, B are iteratively allocated to the units in according to the following steps until the number of bits remaining is less than the minimum number of bits required in the next allocation.

[1] If the remaining bits, B is less than the minimum number of bits required in the next allocation, then the iteration is stopped (step ~~of~~ numeral 6).

[2] Find the unit with the largest variance, var (step ~~of~~ numeral 7).

[3] Increment the bit allocations in the unit by one or two bits (b), WL'[u]=WL[u]+b (step ~~of~~ numeral 8).

[4] Reduce the variance of the unit by var'[u]=var[u]−b*F, where F is the psychoacoustic weighting factor(step ~~of~~ numeral 9).

[5] Recalculate the number of bits remaining, B'=B−b*A[u], where A[u] is the number of spectral lines in the unit, (step ~~of~~ numeral 10).

[6] ~~Step~~ Steps [6] to [10] ~~is~~ are repeated.

The unit, spectrum number and bit allocation ~~data(WL)~~ data (WL) are described as follows.

The audio signal is transformed into ,for example , 256 spectral

lines by the MDCT. The spectral lines are grouped into plural units. In Fig. 6, the 256 spectral lines are grouped into 26 number of units from 0-25. The bit allocation data WL(u) is given by an unit as shown in Fig. 7. In the example of Figs. 6, 7, the spectral line numbers 1-8 have the same bit allocation data WL(0) and the spectral line numbers 9-12 have same bit allocation data WL(1).

Fig. 2 shows the flow chart of the bandwidth computation procedure, a modified set of values of the human hearing threshold, obtained from listening tests, is used to determine the bandwidth of the audio signal and to almost eliminate any units where the variance is below the threshold from the next stage of bit allocation. First, the bandwidth counter is set to zero (step of numeral 31). In a step of numeral 32, the block mode is tested. If the short block mode was used in the MDCT process, the thresholds for units having the same frequency resolution is obtained by selecting the lowest threshold from all these units (step of numeral 33). Each unit's variance is then compared against the threshold (step of numeral 34). If the variance is above the threshold, the bandwidth counter is incremented (step of numeral 35). Otherwise, the variance in the unit is set to a low dummy value so that it will have the least chance of being allocated any bits in the next stage (step of numeral 36). This procedure is repeated for all units (step of numeral 37 and 38). The use of the hearing threshold will

eliminate units that are irrelevant according to the psychoacoustic criteria.

Fig. 3 shows the procedure of allocating the initial bits to each unit before the iterative process by an approximate mathematical model. First, the sum of all the variances and the number of units with variances greater than the low dummy value are calculated (step of numeral 41). Step of numeral 42 checks if all the units within the bandwidth have been processed. If all units are processed, then the variance in each unit is normalized according to the number of bits allocated to the unit (var'[u]=var[u]−WL*F) (step of numeral 44). The procedure is then ended. If all of the units are not processed, then the number of bits to be allocated to the next unit, WL[u] is calculated by equation (1) shown below (step of numeral 43):

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

WL[u] : number of bits allocated to unit u

var[u] : variance of unit u

F : psychoacoustics 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

Steps ~~of~~ numerals 45, 46, 47 and 48 checks if the bit allocations are below the lowest limit or exceeds the highest limit. In either ~~of~~ the case, the lowest limit ~~(0)~~ (0 bits) and the highest limit ~~(15 bits)~~ (15 bits) are respectively.

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 ~~show~~ 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. However, in the higher frequency end, the error spectrum of F = 9.03 is lower and this accounts for the improvement in the noise control at high frequency. This form of allocation considers that strong signals can have greater in-band masking than weak signals and it is more advantageous to vary the error spectrum more closely to the signal spectrum.

The mathematical model is able to allocate ~~at least~~ at least up to 65% of the total number of bits, thus reducing the number of iterations required.

This invention has been simulated to produce very high audio quality when used in the ATRAC algorithm. The bit allocation technique, being computationally less intensive takes up a total of 5.7 Mips. This can contribute significantly to a low cost and low power (due to the low frequency of operation) single chip LSI implementation of the encoder.

WHAT IS CLAIMED IS:

(1) a method of determining a quantization of digital audio signals having spectral and temporal structure wherein the audio signals are buffered in frames and decomposed into its spectral components, which comprises the steps of:

obtaining a variance and/or a representative of the components of the said frame within a defined frequency interval;

computing a necessary bandwidth using a psychoacoustics criteria;

determining an initial quantizations of each said frequency interval using an approximate mathematical model;

iteratively increasing quantizations for satisfying necessary and final bit rate , maintaining effects of said model.

(2) Method according to claim 1, wherein said iteratively increasing quantizations is that :

iteratively increasing quantizations of each said frequency interval through a psychoacoustically weighted inverse relationship with the said variance of each said frequency interval.

(3) Method according to claim 1, wherein

the bandwidth is computed by comparing the said variance or

representative against a human hearing threshold such that the largest frequency interval with said variance or representative exceeding the hearing threshold taken to be the bandwidth.

(4) Method according to claim 1, wherein said mathematical model is derived ~~in consideration of~~ from an in-band masking effect such that an error spectrum will follow more closely to the original signal spectrum.

ABSTRACT

When determining a quantization for ~~of~~ digital audio signals having a spectral and temporal structure wherein the audio signals are buffered in frames and decomposed into ~~its~~ spectral components, the method is: obtaining a variance and/or a representative of the components of the the frame within a defined frequency interval; computing a necessary bandwidth using psychoacoustics criteria; determining an initial quantizations for ~~of~~ each ~~of the~~ frequency interval using an approximate mathematical model; and iteratively increasing the quantization of each ~~quantizations of each~~ the frequency intervals ~~interval~~ through a psychoacoustically weighted inverse relationship with the the variance of each of the frequency intervals ~~interval~~.

Declaration and Power of Attorney For Patent Application

特許出願宣言書

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私は、下欄に氏名を記載した発明者として、以下のとおり宣言する：

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私は、前記のとおり補正した請求の範囲を含む前記明細書の内容を検討し、理解したことを陳述する。

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As a below named inventor, I hereby declare that:

My residence, post office address and citizenship are as stated below next to my name,

I believe I am the original, first and sole inventor (if only one name is listed below) or an original, first and joint inventor (if plural names are listed below) of the subject matter which is claimed and for which a patent is sought on the invention entitled

A bit allocation method for

transform coder

the specification of which

(check one)

☒ is attached hereto.

☐ was filed on _____ as

Application Serial No. _____

and was amended on _____
(if applicable)

I hereby state that I have reviewed and understand the contents of the above identified specification, including the claims, as amended by any amendment referred to above.

I acknowledge the duty to disclose information which is material to the examination of this application in accordance with Title 37, Code of Federal Regulations, §1.56(a).

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Prior foreign applications
先の外国出願

(Number) (番 号)	(Country) (国 名)	(Day/Month/Year Filed) (出願の年月日)	Priority claimed 優先権の主張 Yes あり	No なし
P.appln.hei 5-30,712	Japan	19, February, 1993	☒	☐
			☐	☐
			☐	☐

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I hereby claim the benefit under Title 35, United States Code, §120 of any United States application(s) listed below and, insofar as the subject matter of each of the claims of this application is not disclosed in the prior United States application in the manner provided by the first paragraph of Title 35, United States Code, §112, I acknowledge the duty to disclose material information as defined in Title 37, Code of Federal Regulations, §1.56(a) which occurred between the filing date of the prior application and the national or PCT international filing date of this application:

(Application Serial No.) (出願番号)	(Filing Date) (出願日)	(現 況) (特許済み、係属中、放棄済み)	(Status) (patented, pending, abandoned)

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委任状：私は、下記発明者として、以下の代理人をここに選任し、本願の手続を遂行すること並びにこれに関する一切の行為を特許商標庁に対して行うことを委任する。
（代理人氏名および登録番号を明記のこと）

POWER OF ATTORNEY: As a named inventor, I hereby appoint the following attorney(s) and/or agent(s) to prosecute this application and transact all business in the Patent and Trademark Office connected therewith. (*list name and registration number*)

Joseph A. DeGrandi (17446)
Robert G. Weilacher (20531)
Richard G. Young (20628)
John T. Roberts (20908)
Michael A. Makuch (32263)
Helen M. McCarthy (32513)
Dennis C. Rodgers (32936)
William F. Rauchholz (34701)
G. Byron Stover (34737)

書類の送付先：

Send Correspondence to:

BEVERIDGE, DEGRANDI, WEILACHER & YOUNG
1850 M Street, N.W., Suite 800
Washington, D.C. 20036

直通電話連絡先：（名称および電話番号）

Direct Telephone Calls to: (*name and telephone number*)

BEVERIDGE, DEGRANDI, WEILACHER & YOUNG 659-2811

唯一のまたは第一の発明者の氏名 1.00	Full name of sole or first inventor: Teh, Do Hui
同発明者の署名 日付	Inventor's signature: [signature] Date: 4 Oct 93
住所	Residence: Singapore SGX
国籍	Citizenship: Malaysia
郵便の宛先	Post Office Address: Blk 119 Tampines Court Tampines St.11
	No.12-180 Singapore 1852
第2の共同発明者の氏名（該当する場合） 2.00	Full name of second joint inventor, if any: Tan, Pek Yew
同第2発明者の署名 日付	Second Inventor's signature: [signature] Date: 4 Oct 93
住所	Residence: Singapore SGX
国籍	Citizenship: Malaysia
郵便の宛先	Post Office Address: Blk 128, Yishun St.11
	No.05-305 Singapore 2776

（第六またはそれ以降の共同発明者に対しても同様な情報および署名を提供すること。）

(Supply similar information and signature for third and subsequent joint inventors.)

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第3の共同発明者の氏名（該当する場合） 3.00	Full name of third joint inventor, if any Neo, Sua Hong
同第3発明者の署名 日付	Third Inventor's signature Date 4 Oct 93
住所	Residence Singapore SGX
国籍	Citizenship Singapore
郵便の宛先	Post Office Address Blk 456, Ang Mo Kio Avenue 10
	No.03-1558 Singapore 2056
第4の共同発明者の氏名（該当する場合）	Full name of fourth joint inventor, if any
同第4発明者の署名 日付	Fourth Inventor's signature Date
住所	Residence
国籍	Citizenship
郵便の宛先	Post Office Address
第5の共同発明者の氏名（該当する場合）	Full name of fifth joint inventor, if any
同第5発明者の署名 日付	Fifth Inventor's signature Date
住所	Residence
国籍	Citizenship
郵便の宛先	Post Office Address
第6の共同発明者の氏名（該当する場合）	Full name of sixth joint inventor, if any
同第6発明者の署名 日付	Sixth Inventor's signature Date
住所	Residence
国籍	Citizenship
郵便の宛先	Post Office Address
第7の共同発明者の氏名（該当する場合）	Full name of seventh joint inventor, if any
同第7発明者の署名 日付	Seventh Inventor's signature Date
住所	Residence
国籍	Citizenship
郵便の宛先	Post Office Address

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395/2.38

Fig. 1

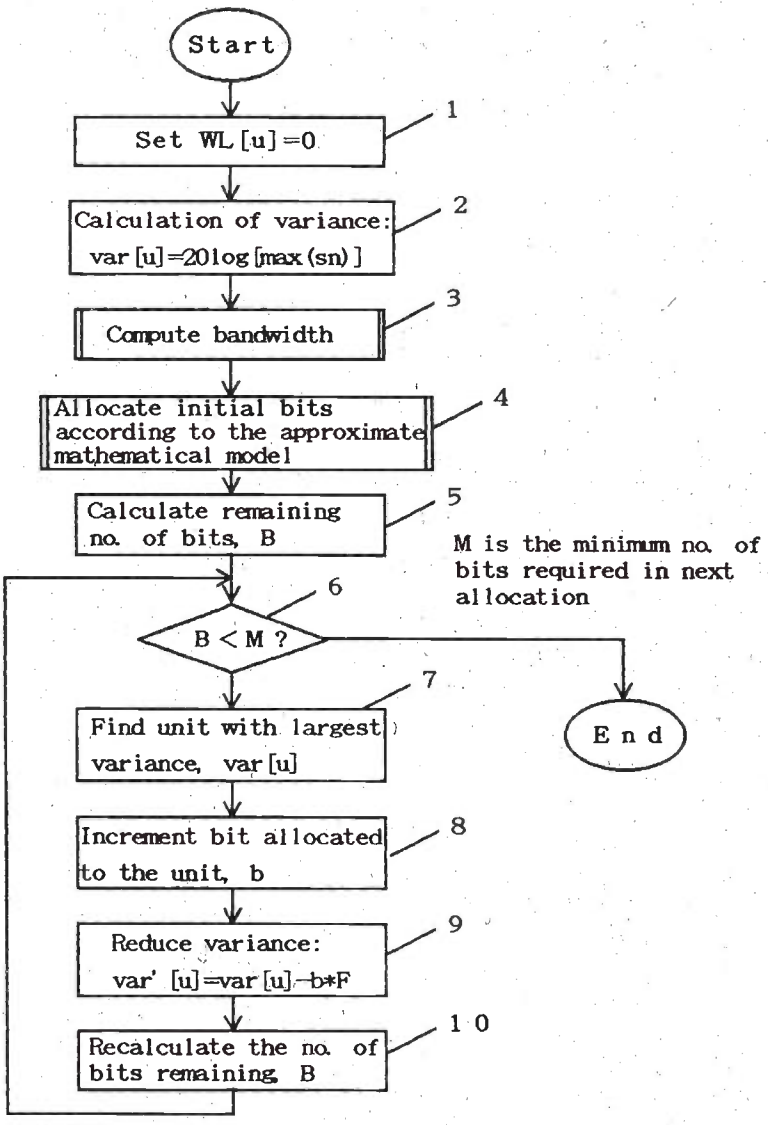
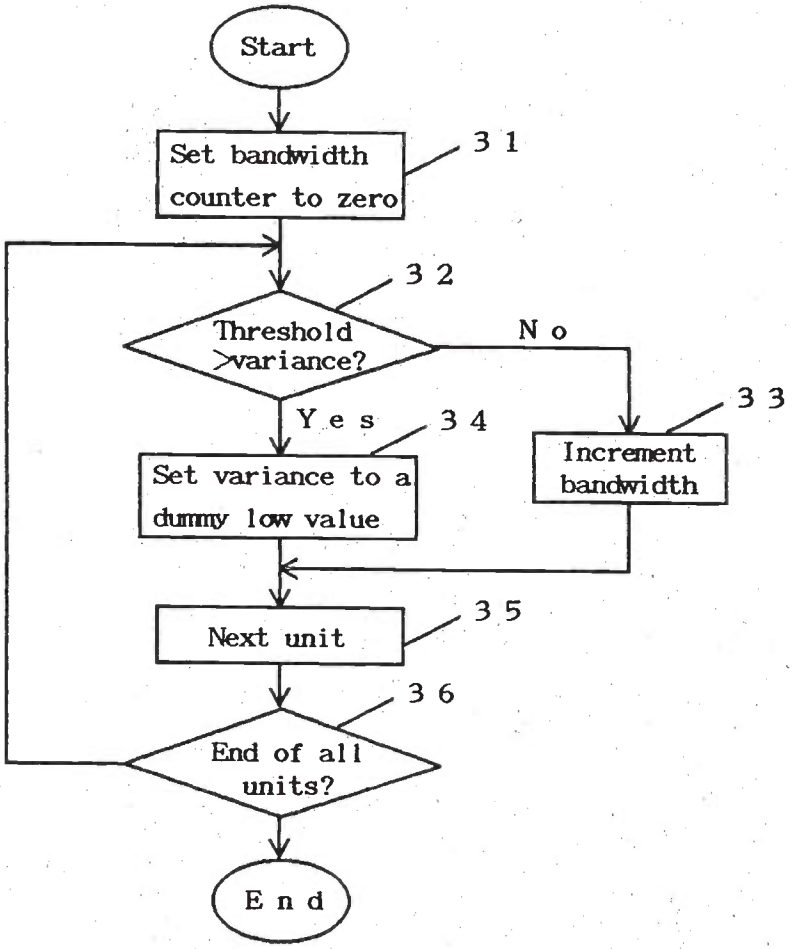
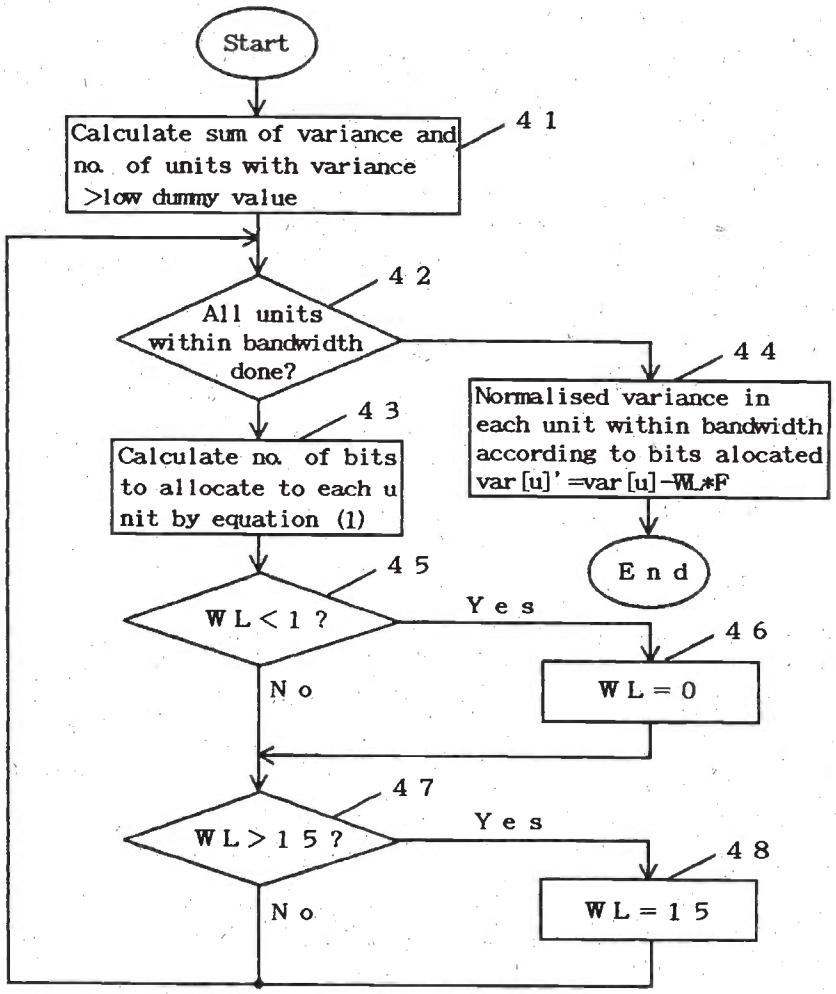


Fig. 2


Start
Set bandwidth counter to zero
31
32
Threshold >variance?
No
Yes
33
34
Increment bandwidth
Set variance to a dummy low value
Next unit
35
36
End of all units?
End

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Fig. 3



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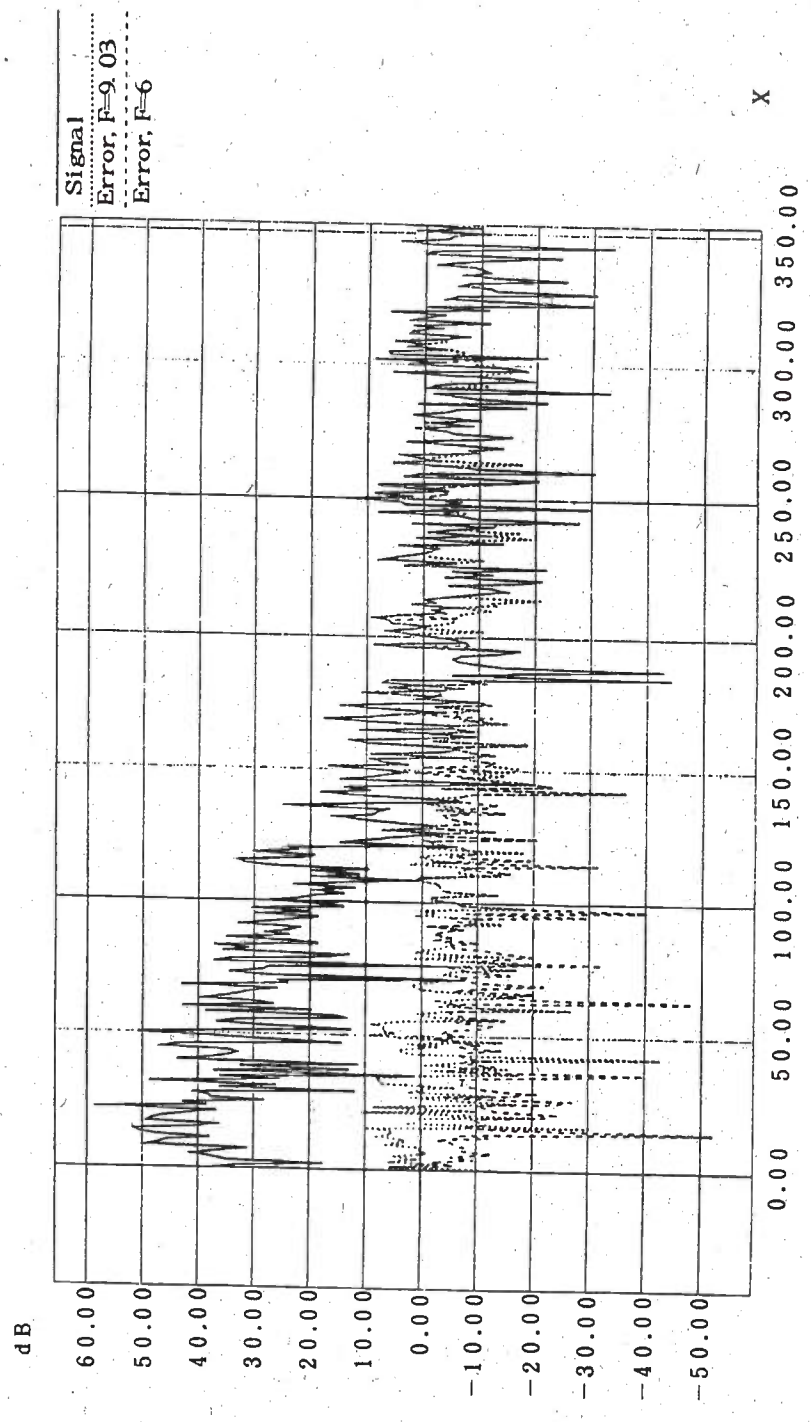
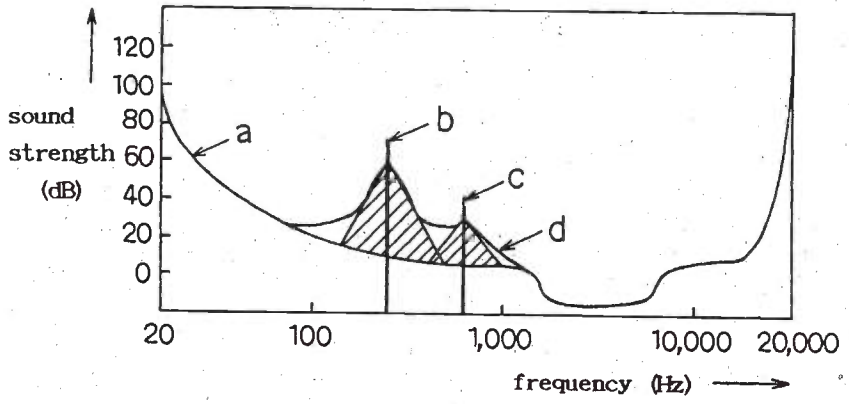


Fig. 4

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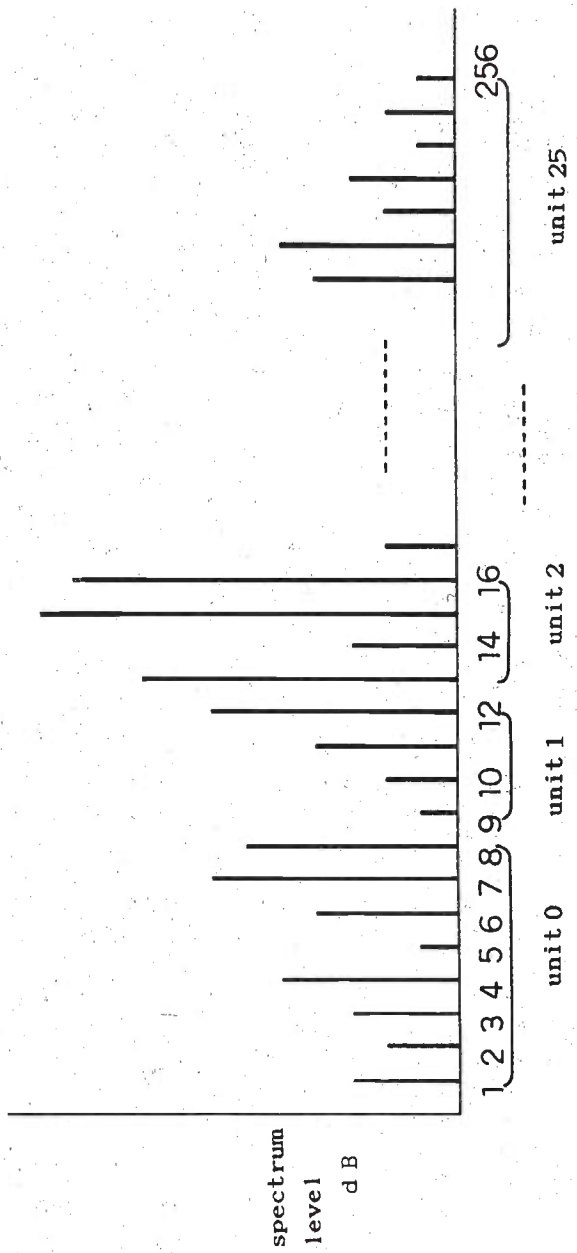
Fig. 5



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Fig. 6



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Fig. 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)

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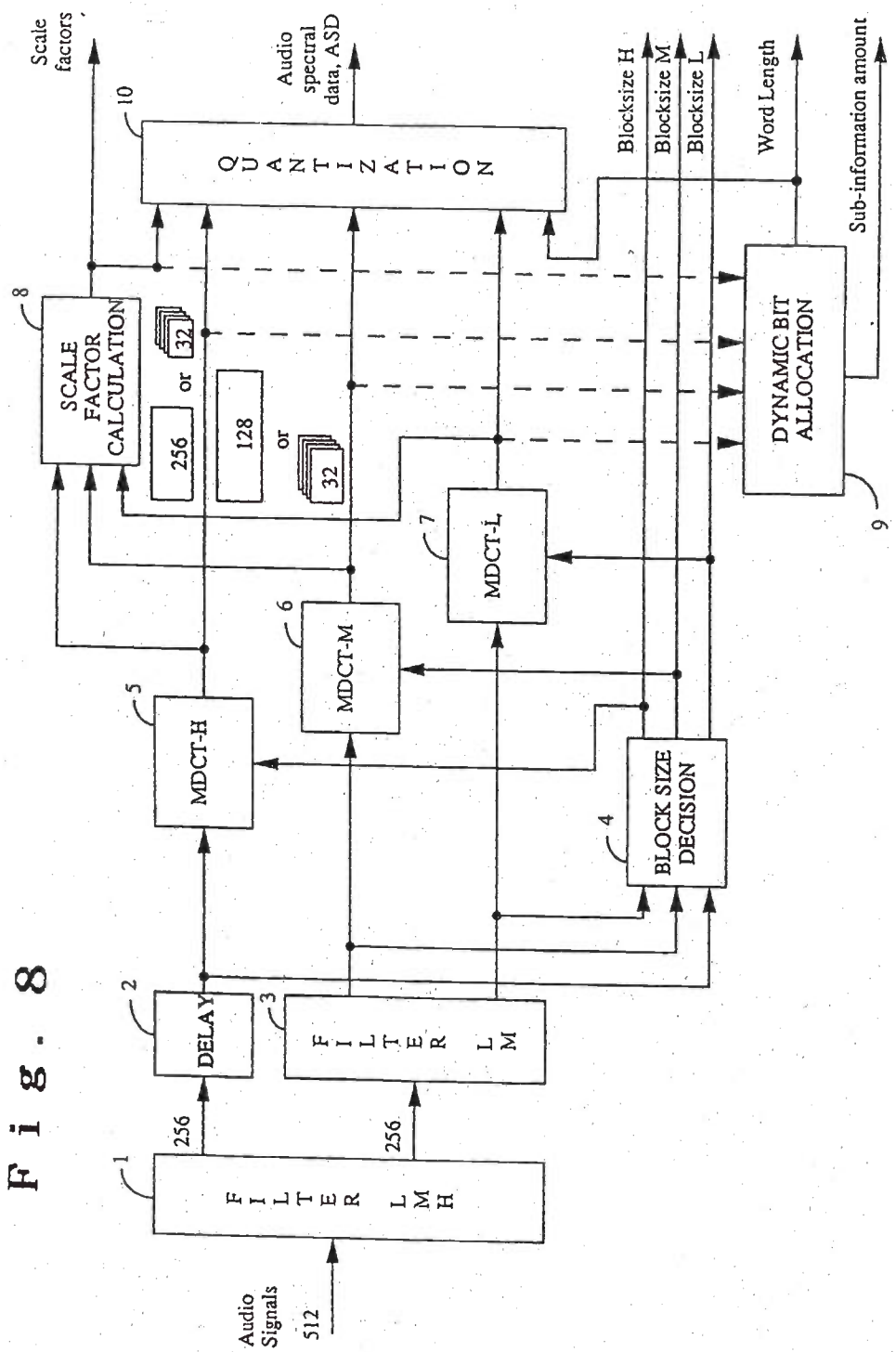


Fig. 8

Fig. 1

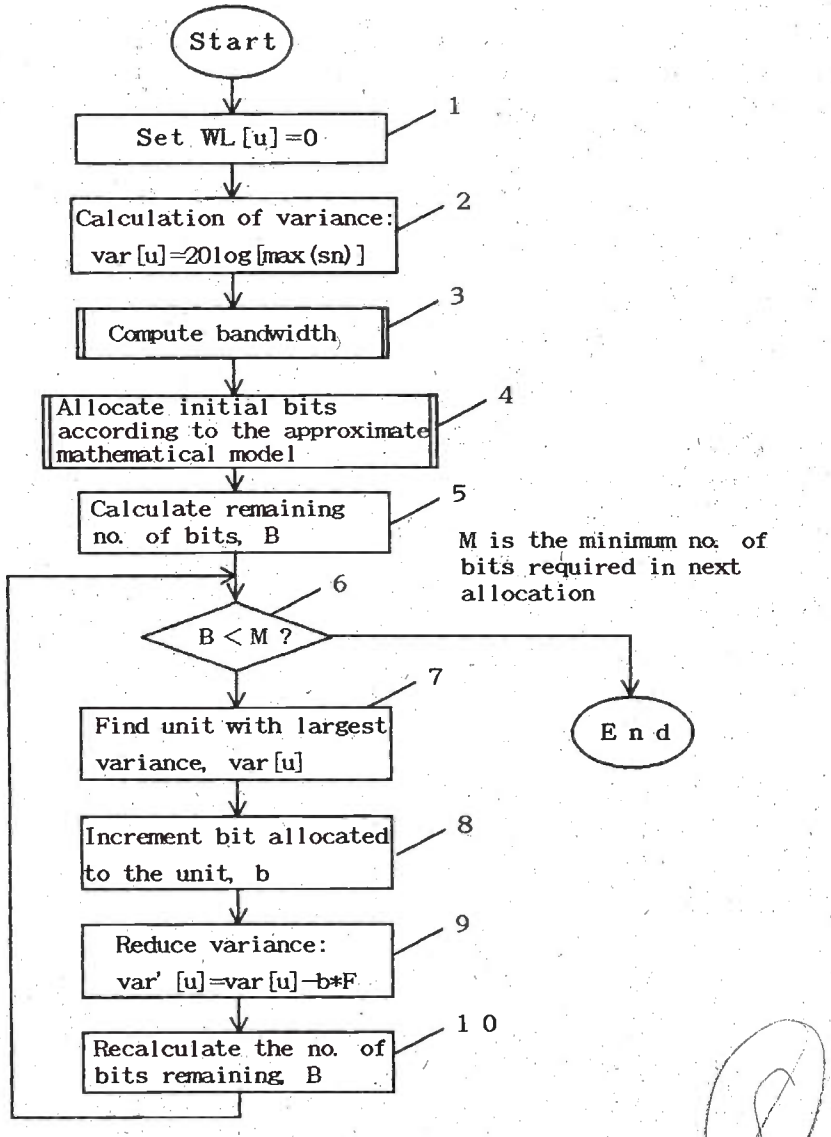

Start
Set WL[u]=0
1
Calculation of variance:
var[u]=20log[max(sn)]
2
Compute bandwidth
3
Allocate initial bits according to the approximate mathematical model
4
Calculate remaining no. of bits, B
5
M is the minimum no. of bits required in next allocation
6
B < M ?
End
7
Find unit with largest variance, var[u]
Increment bit allocated to the unit, b
8
Reduce variance:
var'[u]=var[u]-b*F
9
Recalculate the no. of bits remaining, B
10

Fig. 2

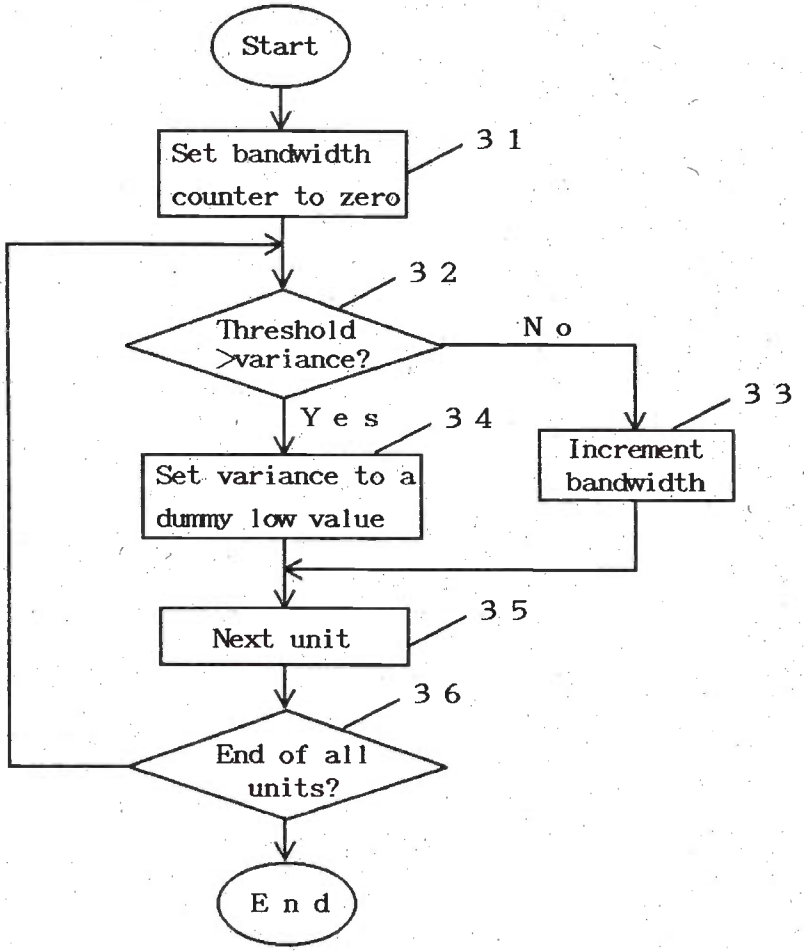

Start
Set bandwidth counter to zero
31
32
Threshold >variance?
No
Yes
33
34
Increment bandwidth
Set variance to a dummy low value
Next unit
35
36
End of all units?
End

Fig. 3

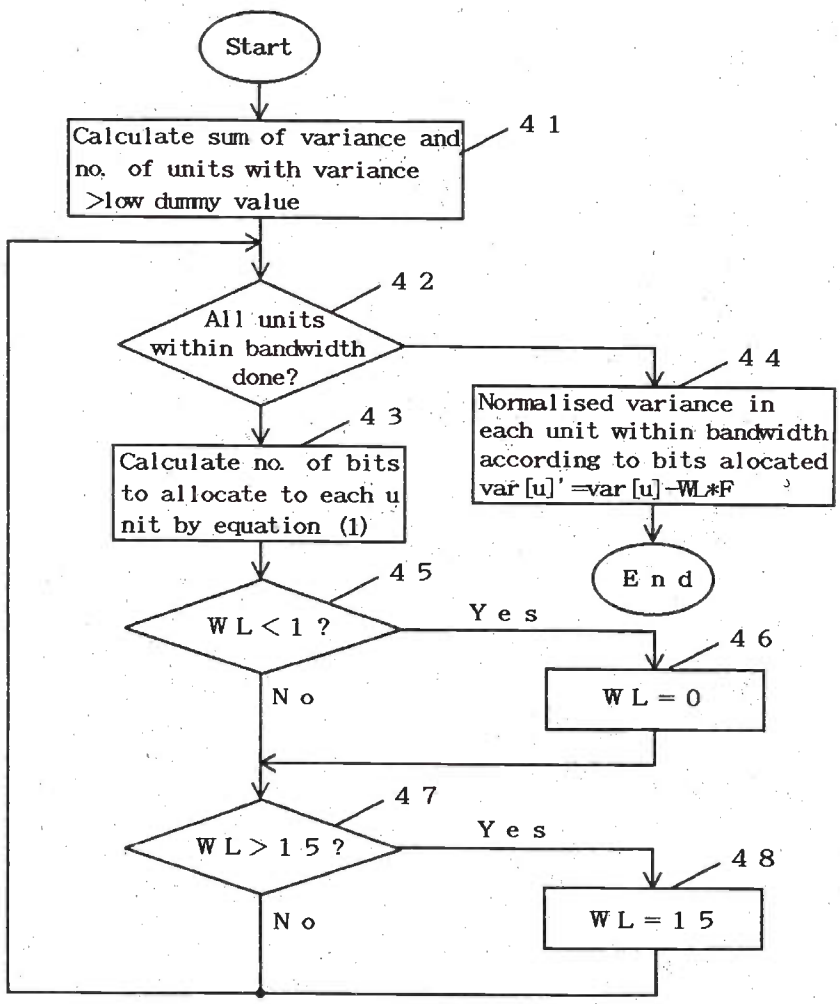

Start
Calculate sum of variance and no. of units with variance >low dummy value
41
All units within bandwidth done?
42
Calculate no. of bits to allocate to each unit by equation (1)
43
Normalised variance in each unit within bandwidth according to bits alocated var[u]' =var[u]-WL*F
44
End
WL<1 ?
45
Yes
No
WL=0
46
WL>15 ?
47
Yes
No
WL=15
48

Fig. 4

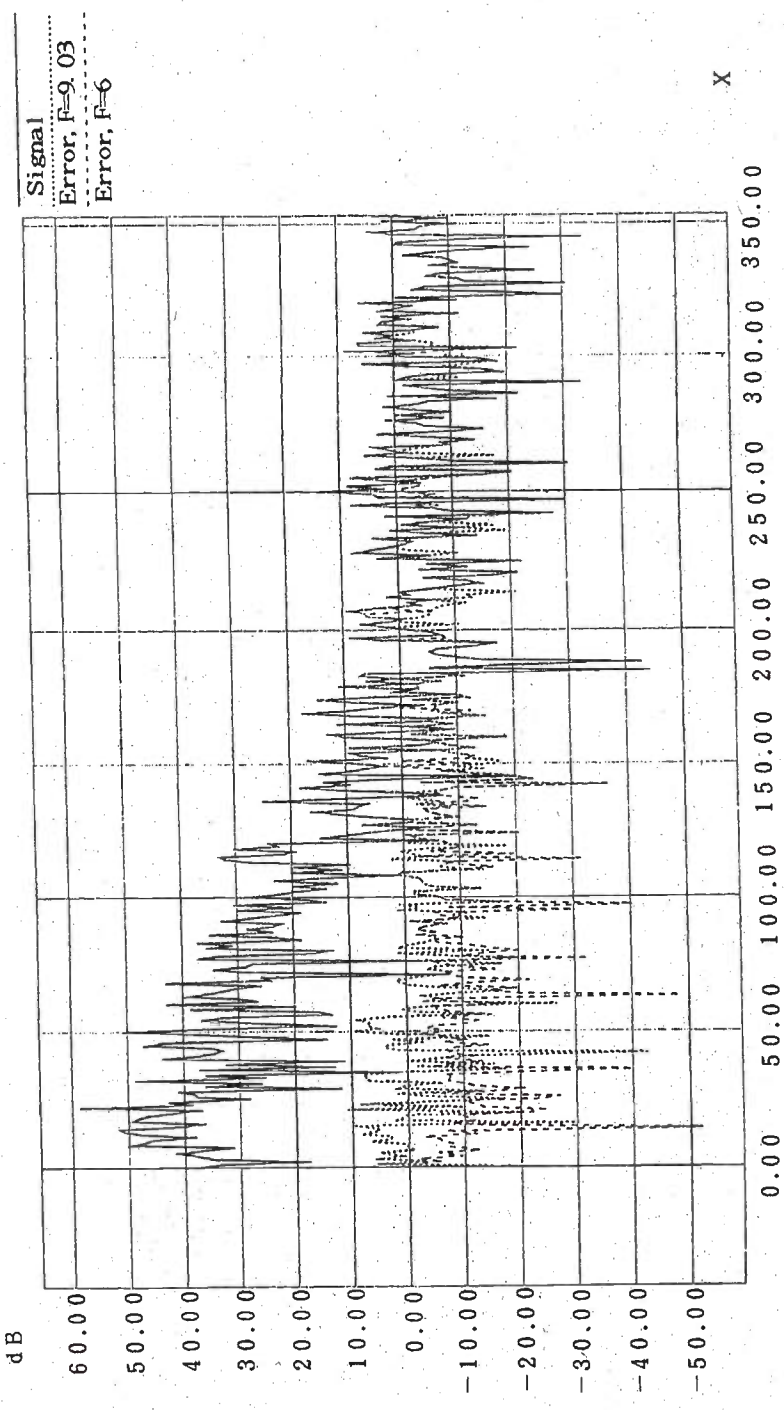

Signal
Error, F=9.03
Error, F=6
dB
60.00
50.00
40.00
30.00
20.00
10.00
0.00
-10.00
-20.00
-30.00
-40.00
-50.00
0.00
50.00
100.00
150.00
200.00
250.00
300.00
350.00
X

Fig. 5

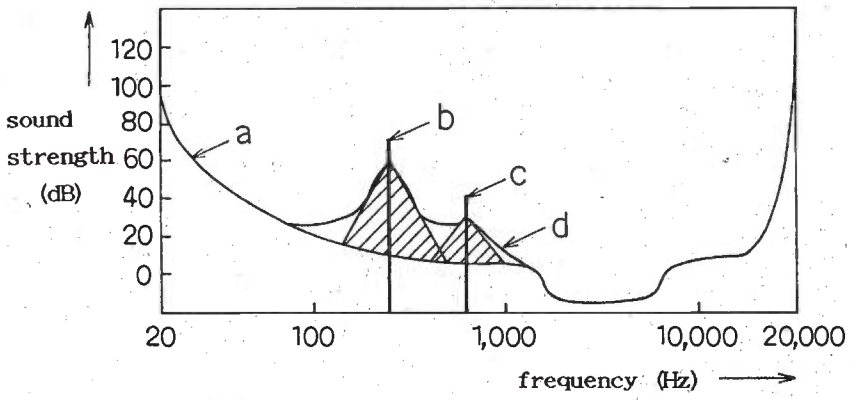

120
100
80
60
40
20
0
sound
strength
(dB)
a
b
c
d
20
100
1,000
10,000
20,000
frequency (Hz)

Fig. 6

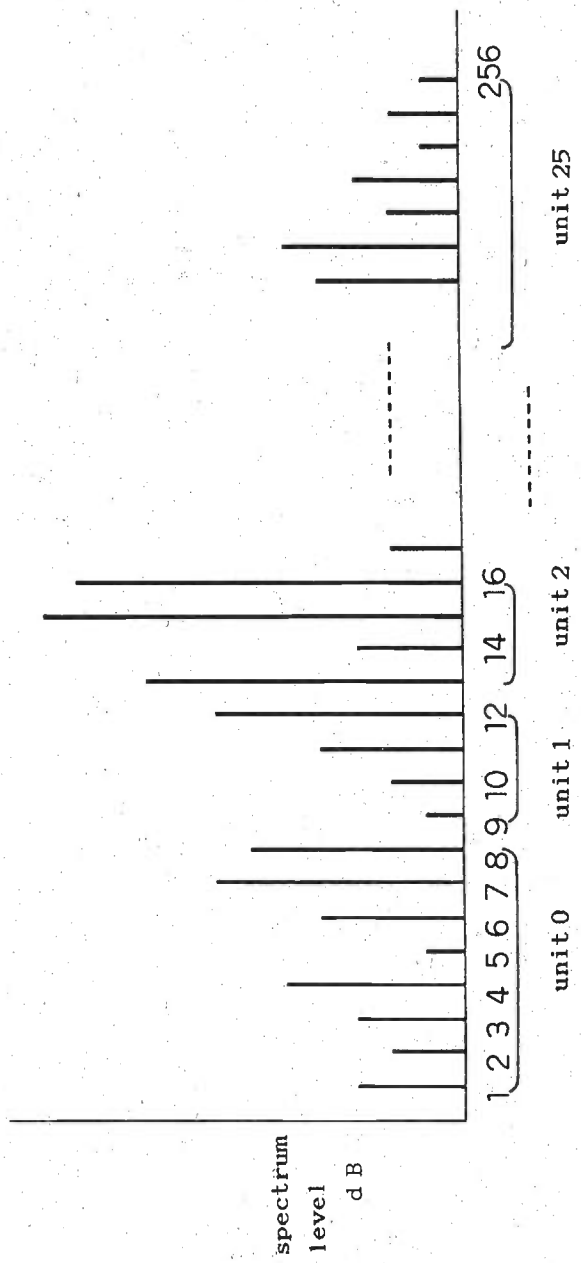
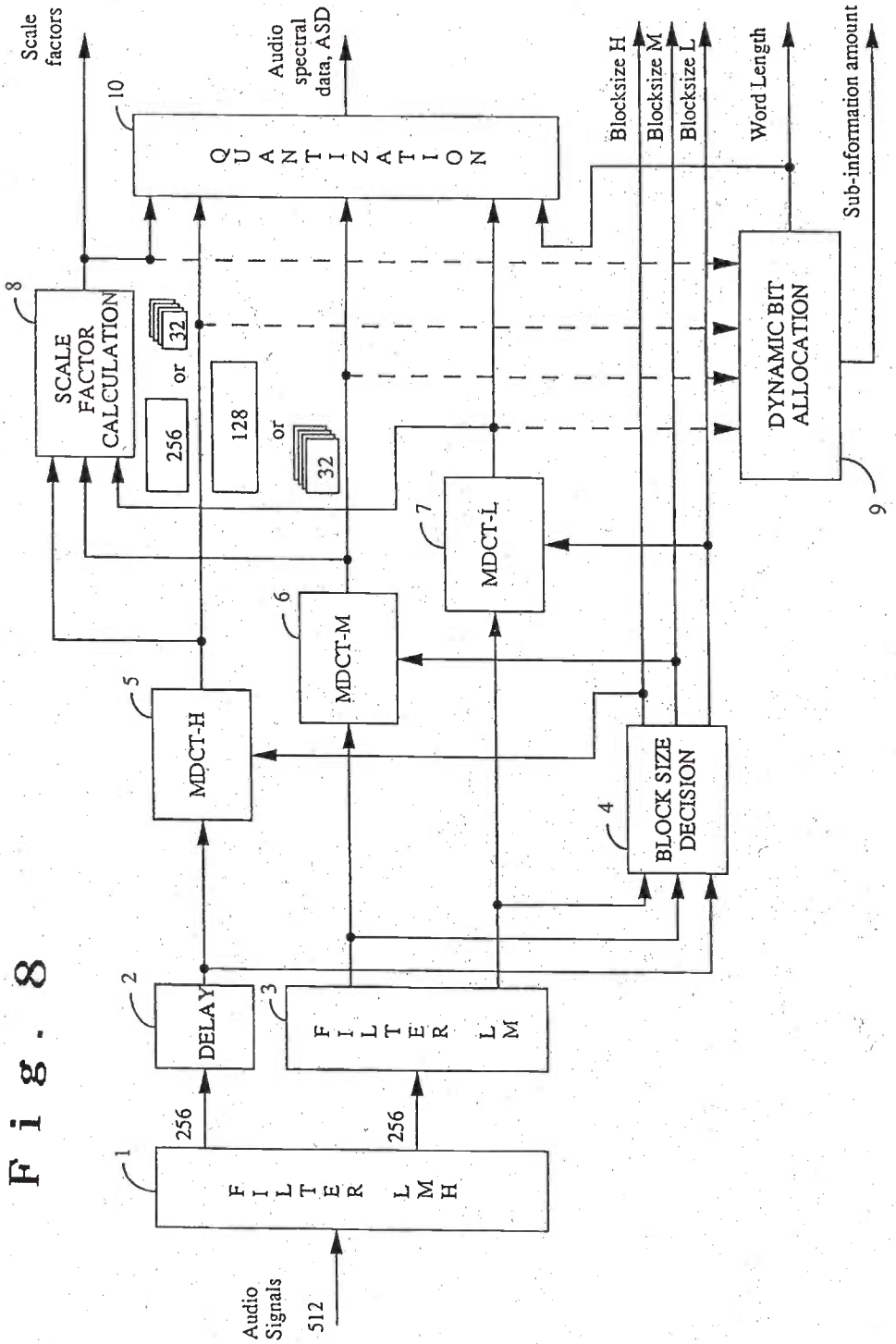

spectrum
level
dB
1 2 3 4 5 6 7 8 9 10 12 14 16
256
unit 0
unit 1
unit 2
unit 25

Fig. 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)

Fig. 8


Audio Signals
512
FILTER LMH
1
256
256
DELAY
2
FILTER LM
3
BLOCK SIZE DECISION
4
MDCT-H
5
MDCT-M
6
MDCT-L
7
SCALE FACTOR CALCULATION
8
DYNAMIC BIT ALLOCATION
9
QUANTIZATION
10
256
or
32
128
or
32
Scale factors
Audio spectral data, ASD
Blocksize H
Blocksize M
Blocksize L
Word Length
Sub-information amount



Inventors: TEH, et al.

Serial No.: To be assigned

Filing Date: December 6, 1993

Sheet 8 of 8

Application Assignment Record

According to the application transmittal letter, an assignment recording ownership was filed with this application; however, a copy of this record was not located in the original file history record obtained from the United States Patent and Trademark Office. Upon your request, we will attempt to obtain the assignment documents from the Assignment Recordation Branch of of the United States Patent and Trademark Office or from a related application case (if applicable). Please note that additional charges will apply for this service.

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03-48683



PATENT

#2 Waiver 2-22-94

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicants: TEH, et al.

Serial No.: To be assigned

Filed : December 6, 1993

For : A BIT ALLOCATION METHOD FOR TRANSFORM CODER

CLAIM FOR PRIORITY

Hon. Commissioner of Patents
and Trademarks
Washington, D.C. 20231

Sir:

Under 35 U.S.C § 119, Applicants claim the benefit of the filing date of Patent Application 5-030,712, filed in Japan on February 19, 1993.

In support of this claim, a certified copy of the above-mentioned application is attached.

Respectfully submitted,

BEVERIDGE, DeGRANDI, WEILACHER & YOUNG

By: Joseph A. DeGrandi

Joseph A. DeGrandi, Reg. No. 17,446
1850 M Street, N.W., Suite 800
Washington, D.C. 20036
Telephone: (202) 659-2811

December 6, 1993

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PATENT OFFICE
JAPANESE GOVERNMENT

別紙添付の書類に記載されている事項は下記の出願書類に記載されている事項と同一であることを証明する。
This is to certify that the annexed is a true copy of the following application as filed with this Office.

出願年月日
Date of Application: １９９３年　２月１９日

出願番号
Application Number: 平成　５年特許願第０３０７１２号

出願人
Applicant (s): 松下電器産業株式会社

１９９３年　５月１４日

特許庁長官
Commissioner,
Patent Office

麻生渡

出証平　０５－０２１７９６

０５－０３０７１２

【書類名】　特許願
【整理番号】　2054050003
【提出日】　平成 5年 2月19日
【あて先】　特許庁長官殿
【国際特許分類】　H03M 7/30
【発明の名称】　変換符号化装置のビット配分方法
【請求項の数】　4
【発明者】
【住所又は居所】　シンガポール　１８５２　＃１２－１８０　タンピンズ　ストリート１１　タンピンズ　コート　ブロック１１９
【氏名】　テ　ドゥ　ウィ
【発明者】
【住所又は居所】　シンガポール　２７７６　＃０５－３０５　イシュンストリート１１　ブロック１２８
【氏名】　タン　ペク　ユー
【発明者】
【住所又は居所】　シンガポール　２０５６　＃０３－１５５８　アンモキオ　アベニュー１０　ブロック４５６
【氏名】　ニオ　スァ　ホング
【特許出願人】
【識別番号】　000005821
【郵便番号】　571
【住所又は居所】　大阪府門真市大字門真１００６番地
【氏名又は名称】　松下電器産業株式会社
【代表者】　谷井　昭雄
【代理人】
【識別番号】　100092794
【郵便番号】　532

０５－０３０７１２

【住所又は居所】　大阪市淀川区宮原五丁目１番３号新大阪生島ビル５Ｆ
【弁理士】
【氏名又は名称】　松田　正道
【電話番号】　06 397-2840
【手数料の表示】
【納付方法】　予納
【予納台帳番号】　009896
【納付金額】　14,000円
【提出物件の目録】
【物件名】　明細書　1
【物件名】　図面　1
【物件名】　要約書　1
【包括委任状番号】　9006027
【プルーフの要否】　要

【書類名】　明細書

【発明の名称】　変換符号化装置のビット配分方法

【特許請求の範囲】

【請求項１】　オーディオ信号がフレーム内でバッファされ、そのスペクトル成分に分解される、スペクトル及び時間的構造を有するディジタル・オーディオ信号の量子化を決定する方法において、

所定の周波数間隔内の前記フレームの成分の分散及び／又は代表値を得、

心理音響学的基準を用いて必要な帯幅を計算し、

信号の帯域内マスキング効果を考慮する粗いモデルを用いて前記周波数間隔の各々の初期量子化を決定し、

前記モデルの効果を維持しつつ必要な最終的なビット速度を満たすため量子化を反復的に増加する、

ことを特徴とする変換符号化装置のビット配分方法。

【請求項２】　モデルの効果を維持しつつ必要な最終的なビット速度を満たすため量子化を反復的に増加するプロセスは、前記周波数間隔の各々の前記分散との心理音響学的重み逆関係を通して前記周波数間隔の各々の量子化を反復的に増大するものであることを特徴とする請求項１記載の変換符号化装置のビット配分方法。

【請求項３】　聴覚しきい値を超過する分散及び／又は代表値の最大周波数間隔を帯幅として取るようにして、人間の聴覚しきい値に対して前記分散及び／又は代表値を比較して帯幅が計算されることを特徴とする請求項１記載の変換符号化装置のビット配分方法。

【請求項４】　前記モデルは、誤差スペクトルはオリジナルの信号スペクトルにより密接に従うという帯域内マスク効果を考慮して導出されることを特徴とする請求項１記載の変換符号化装置のビット配分方法。

【発明の詳細な説明】

【０００１】

【産業上の利用分野】

本発明は、送信あるいはディジタル記憶媒体用のディジタル・オーディオ信号

の効率的な情報符号化、特にその変換符号用の効率的なビット割当手法を用いる量子化方法に関するものである。

【０００２】

【従来の技術】

最近、広帯域オーディオ信号の圧縮アルゴリズムが、広く適用されるようになってきている。家電分野では、ディジタル・コンパクト・カセット（ＤＣＣ）とミニ・ディスク・システム（ＭＤ）の２つの分野で、それらの圧縮アルゴリズムが適用されている。ＤＣＣシステムは副帯域符号化に基づいた圧縮アルゴリズムを使用するが、ＭＤシステムは副帯域と変換符号化のハイブリッドに基づくアルゴリズムを使用し、変換符号化部分がアルゴリズムのバックボーンを形成している。本発明はＭＤシステムで使用する変換符号器のブロック・サイズの判定に関連したものである。

【０００３】

ＭＤシステムは「ＡＴＲＡＣ(ADAPTIVE TRANSFORM ACOUSTIC CODING)圧縮アルゴリズム」を使用する。このＡＴＲＡＣアルゴリズムは、７０５．６ｋｂｉｔ／ｓ／チャネルのビット速度の入力オーディオ信号を１４６．０８ｋｂｉｔ／ｓ／チャネルのビット速度に圧縮するものである。

【０００４】

図８にその符号化過程を実現するブロック図を示す。入力時間信号はまず分割フィルターバンク１、２、３を通過して、３つの周波数帯域の信号を得る。低い方の２つの帯域はそれぞれ最も高い帯域の帯幅の半分である。ブロックサイズ決定過程４は各々の帯域について行われ、ウィンドウ化と変換過程５、６、７のためにサンプルサイズないしブロックモードを決定する。ＭＤＣＴ－Ｈ５、ＭＤＣＴ－Ｍ６，ＭＤＣＴ－Ｌ７は、修正ディスクリートコサイン変換手段であって、時間領域のディジタルオーディオ信号を周波数領域のスペクトルデータに変換するものである。以降、修正ディスクリートコサイン変換をＭＤＣＴと略記する。その時間単位がブロックサイズ決定過程４で決められる。そして、利用し得る短ブロックモードないし長ブロックモードの得られる２つのブロックモードの１つが帯域の各々について選択される。変換されたスペクトル・サンプルはユニット

にグループ化され、各々のユニットでは、ユニット８のサンプルのピーク値から換算係数を導出する。それらのユニットは非均一な周波数間隔で、低周波数では分解度は精細で、高周波数では分解度は粗である。量子化１０は、換算係数と動的ビット割当モジュール９からのビット割当情報を用いてサンプルに対して行う。

【０００５】

この動的ビット割当方法は、どの様な適応的圧縮アルゴリズムでも不可欠の部分となっている。再構成した出力の質と冗長性と無関係性除去の程度は、ビット割当方法により大きく左右される。更に、ビット割当手順はハードウエアの複雑性の度合をも決定する役割を果たす。ＡＴＲＡＣアルゴリズムでのビット割当はおもに変換スペクトルサンプルに適用されるので、かなりの数の従来技術がこの種の変換符号プロセスについて存在する。それらの動的ビット割当手法は、大きく分けて２つのカテゴリーにグループ化することができる。

【０００６】

第１のカテゴリーは、マスキングしきい値を導出するため同時マスキング及び静寂でのしきい値という心理音響学的現象を適用するビット割当方法からなる。このカテゴリーの例としては、ＭＤＣＴスペクトルサンプルを用いてマスキングしきい値を計算する。より複雑な手法としては、高速フーリエ変換（ＦＦＴ）を用いてより複雑なマスキング計算の周波数スペクトル成分を得るJDジョンストンによる「感覚的雑音基準を用いたオーディオ信号の変換符号化」という名称の論文に記載されている。

【０００７】

第２のカテゴリーは、信号統計学にしたがって割り当てるという、より単純な方法を提供する。例としてNS．ジャヤントとP.ノルによる「波形のディジタル符号化」に記述された最適ビット割当手順があり、一定のビット速度に対する再構成した誤差を最小にすることによりビットを異なるスペクトル成分に割り当てている。

【０００８】

【発明が解決しようとする課題】

ところで、動的ビット割当手順の目的は、オリジナルのオーディオ品質を維持しつつ冗長性と無関係性を除去することである。一般に低価格の消費者製品でこのアルゴリズムの適用を可能にするため、アルゴリズムは簡単なものでなければならない。動的ビット割当手順の設計での最大の問題は、手順をできるだけ単純にしながら同時に良好なオーディオ品質のバランスを取ることにある。先にカテゴリー１の下で記述したビット割当手順は、良好な音質を生成することができるが、かなりの複雑性が存在する。それらの大部分のアルゴリズムはビット割当を行うのに強力なＤＳＰの（固定小数点ないし浮動小数点で）処理能力を必要とする。

【０００９】

他方、上述の最適ビット割当手順は複雑性は少ないが、その設計は純粋に出所源、即ちオーディオ信号に集中しており、最終的なレシーバ即ち人間の聴覚システムではないので、感覚ベースのビット割当の品質を達成することはできない。

【００１０】

更に割当手順では、チャネル・ビット速度を最適に使い尽くすために多数の反復ループを必要とする。この問題は「波形のディジタル符号化」の著者により明らかにされた。従ってこのビット割当アルゴリズムはＤＳＰの実行時間に対して多くの制約がある場合は、不適切である。この集中的な反復の問題を倍加するものとして、ＭＤ符号化アルゴリズムでは、非均一の数のスペクトルサンプルが各々のユニットにグループ化され、従ってビット割当でビット無しに使用されるビットの数を計算するのは困難になるという事実がある。

【００１１】

本発明は、このような従来のビット割当の課題を考慮し、低コスト目標を満たしつつ、高オーディオ品質を達成できるように、低レベルの複雑性で人間の聴覚システムに適応するようにした変換符号化装置のビット配分方法を提供することを目的とするものである。

【００１２】

【課題を解決するための手段】

本発明は、オーディオ信号がフレーム内でバッファされ、そのスペクトル成分

に分解されるスペクトル及び時間的構造を有するディジタル・オーディオ信号の量子化を決定する際、所定の周波数間隔内の前記フレームの成分の分散及び／又は代表値を得て、心理音響学的基準を用いて必要な帯幅を計算し、すなわち無関係性を除去するため人間の聴覚のしきい値を用いてオーディオ信号の必要な帯幅を判定し、信号の帯域内マスキング効果を考慮する粗いモデルを用いて前記周波数間隔の各々の初期量子化を決定し、モデルの効果を維持しつつ必要な最終的なビット速度を満たすため量子化を反復的に増加する方法である。

【００１３】

【作用】

ビット割当アルゴリズムは再構成出力の品質とハードウエア実現の複雑性を決定する際に非常に重要な役割を果たす。本発明では低ハードウエア複雑性で高品質オーディオを生成する動的ビット割当方式を、マスキングの心理音響学的現象と聴覚しきい値を結合させることにより実現した。ハードウエアが簡単な点は、近似数学的モデルを使用することにより精細な反復過程を用いて残りのビット数を割り当てる前に初期ビット数を割り当てることで達成する。この低複雑性ビット割当手順により単一チップ・エンコーダの実現が可能になる。

【００１４】

すなわち、本発明では、ビット割当手順で使用する分散ないし代表値は、信号の動的挙動を考慮しそれに対して調節を行うのに対して、帯幅計算は人間の聴覚システムを考慮し、ビット割当をそれに適応させる。モデルもビット割当に必要な計算ステップ数ないしループを削減するようにしつつマスキングを考慮することにより人間の聴覚システムに適応する。これは統計的に導出された定数との関係内で動的要素を近似することで手順を更に単純化することを意味する。反復手段により、モデルで作られた近似に対して調節しつつ所望のビット速度を確実に満たすことができる。

【００１５】

ここで、マスキングのしきい値に付いて説明する。図５は、マスキングのしきい値の例を示す。図に於て、ａは最小可聴値を示し、この曲線より上の範囲が人間の可聴域である。また、ｂとｃとは、強勢な信号成分のスペクトルを示し、斜

線を施した部分は、信号ｂとｃによってマスクされる部分である。曲線ｄは可聴域と信号ｂ，ｃによってマスクされる斜線部分とから得られるマスキングのしきい値である。このしきい値ｄ以下の信号は感知されないので、量子化する必要はない。また、このしきい値以上の信号であっても、量子化雑音をこのしきい値以下のなるように量子化すれば、人間の耳には量子化雑音は感知されない。従って、量子化のビット配分の決定は、このしきい値と信号成分の大きさとの関係において、決定される。

【００１６】

【実施例】

以下、本発明の実施例について図面を参照して説明する。

【００１７】

図１は本発明の一実施例にかかる変換符号か装置のビット配分方法を示すフローチャートである。最初に各々の周波数間隔又はユニット（ＷＬ［ｕ］）に対する量子化又はビット割当をゼロに設定する（ステップ１）。ユニットの詳細は、次のパラグラフで説明する。各々のユニットの分散は、ユニット内の最大スペクトルサンプル振幅から計算する（ステップ２）。本発明の他の実施例としては、ステップ２において、分散をユニット内のスペクトル成分の平均から計算することもできる。最大スペクトルサンプル振幅は、それがトーン状あるいは雑音のようであってもユニット内の信号の良好な最悪ケースの代表値であることが分かった。更に最大スペクトルサンプルを使用することで同じ計算ステップを省略することができる。送信の帯幅を次に計算する（ステップ３）。ステップ４で、ビットの初期の数を近似数学的モデルにしたがって割り当てる。すなわち、粗く割り当てられる。残りのビットの計算はそこで、ステップ５で行う。残りのビットのＢは、残っているビットの数が次の割当で必要な最低ビット数以下になるまで以下のステップにしたがって、反復的にユニットに割り当てられる。

すなわち、

[1] 残りのビットのＢが次の割当で必要な最低ビット数よりも少なければ、反復を停止する（ステップ６）。

[2] 最大の分散varを有するユニットを求める（ステップ７）。

[3] ユニット内のビット割当を１ないし２ビット（ｂ）だけ増分する。

【００１８】

$$WL'[u] = WL[u] + b \quad \text{（ステップ８）}$$

[4] ユニットの分散を var'[u] = var[u] - b*F だけ減少する。ここでＦは心理音響学的重み係数である（ステップ９）

[5] 残りのビットの数を再計算する（B' = B - b* A[u]）。ここでＡ［ｕ］はユニット内のスペクトル線の数である（ステップ１０）。

[6] ステップ１からステップ５を繰り返す。

【００１９】

ここで、上記ユニット、スペクトル本数及びビット配分データ（ＷＬ）に付いて説明する。

【００２０】

オーディオ信号はＭＤＣＴによって、例えば２５６本の周波数スペクトルに変換される。周波数スペクトルは、複数のユニットにグループ化される。ユニットの例を図６に示す。図６では、例えば２５６本のスペクトルを０から２５の２６個のユニットにグループ化している。ビット配分データＷＬ（ｕ）は、図７に示すように、ユニット単位で与えられる。図６及び図７の例では、１番目から８番目のスペクトルは、同じビット配分データＷＬ（０）を共有し、９番目～１２番目のスペクトルは、ＷＬ（１）を共有する。

【００２１】

図２は、帯幅計算手順の流れ図である。聴覚テストで得た人間の聴覚のしきい値の１組の修正値を用いてオーディオ信号の帯幅を判定し、ビット割当の次の段階から分散がしきい値以下のユニットをほとんど除去する。まず、帯幅カウンタをゼロにセットする（ステップ３１）。次に、各々のユニットの分散をしきい値に対して比較する（ステップ３２）。分散がしきい値以上であれば、帯幅カウンタを増分する（ステップ３３）。そうでなければユニット内の分散は次の段階でビットが割り当てられることがないように低ダミー値にセットされる（ステップ３４）。この手順は全てのユニットについて繰り返される（ステップ３５、３６）。聴覚しきい値を使用することで心理音響学基準からみて無関係なユニットを

除去する。

【００２２】

図３は反復過程前に近似数学的モデルにより初期ビットを各々のユニットに割り当てる手順を示したものである。まず全ての分散の合計と低ダミー値よりも大きい分散を有するユニットの数を計算する（ステップ４１）。ステップ４２で帯幅内の全てのユニットが処理されたかどうかを調べる。全てのユニットが処理されれば、各々のユニット内の分散はユニットに割り当てられたビットの数にしたがって正規化される（var'[u] = var[u] - WL*F）（ステップ４４）。手順はそこで終了する。全てのユニットが処理されていなければ、次のユニットに割り当てられるビットの数のＷＬ［ｕ］を次の（式１）により計算する（ステップ４３）。

【００２３】

【数１】

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

【００２４】

WL[u]　：ユニットｕに割り当てたビットの数

var[u]　：ユニットｕの分散

F　　：心理音響学的重み係数（定数）

N　　：有効ユニットの数

Ba　　：初期割当のために別の設定されたビットの数

A　　：ユニット内のスペクトルサンプルの近似平均数

ステップ４５、４６、４７、４８では、ビット割当が最低限度以下であるかあるいは最高限度を超過するかを調べる。どちらの場合も最低限度（０）と最高限度（１５ビット）がそれぞれ設定される。

【００２５】

数学的モデルは、生成された誤差スペクトルは、信号スペクトルに、より密接にしたがうようにビットを割り当てるという圧伸効果を持っている。これはＦの値を８．５と１０の間に設定することにより達成される。Ｆ＝６の時、単調な誤

差スペクトルが得られ、これはいくつかのテスト・シーケンスについて作動ビット速度で高周波の人工的雑音を生成することが分かった。

【００２６】

図４はバイオリン信号のフレームのスペクトルプロット、Ｆ＝９．０３の時の誤差スペクトル、Ｆ＝６の時の誤差スペクトルを示したものである。この図から分かるように、Ｆ＝９．０３誤差スペクトルは低周波数では高い誤差値を示すが、それらの誤差は帯域内の遥かに強力な信号によりマスクされるので目立たない。しかし高い周波数端では、Ｆ＝９．０３の誤差スペクトルは低くなり、これは高周波での雑音制御の向上の原因となる。この割当の形式は、強力な信号は弱い信号よりも大きな帯域内マスキングを持つことができることを意味し、誤差スペクトルを信号スペクトルに、より近く変化させることがより有利であることがわかる。数学的モデルは少なくとも６５％の合計ビット数まで割り当てることが出来、従って必要な反復の数を削減できる。

【００２７】

以上の実施例で、ビット割当手法は計算的には余り集中しておらず、合計５．７Ｍｉｐｓを使用する。これはエンコーダの低コスト、低電力（低い作動周波数による）単一チップＬＳＩの実現にかなり貢献するものである。

【００２８】

【発明の効果】

以上述べたところから明らかなように、本発明をＡＴＲＡＣアルゴリズムに適用することによって、非常に高いオーディオ品質を実現できる。

【００２９】

又、低コスト、低電力に貢献できるものである。

【図面の簡単な説明】

【図１】

本発明の一実施例にかかる変換符号化装置のビット配分方法の流れ図である。

【図２】

本実施例における帯幅計算プロセスの流れ図である。

【図３】

本実施例における初期ビット割当プロセスの流れ図である。

【図４】

バイオリン信号のフレームのスペクトルプロットとＦの異なる値で得た誤差スペクトルを示すグラフである。

【図５】

マスキング効果を説明するためのグラフであって、横軸は周波数、縦軸は音圧でを示す。

【図６】

本実施例におけるユニットとスペクトルレベルとの関係を示すグラフである。

【図７】

本実施例におけるユニット番号とスペクトル本数とビット配分データとの関係を示すグラフである。

【図８】

符号化過程のブロック図である。

【符号の説明】

１：フィルタＬＭＨ

２：遅延

３：フィルタＬＭ

４：ブロックサイズ決定

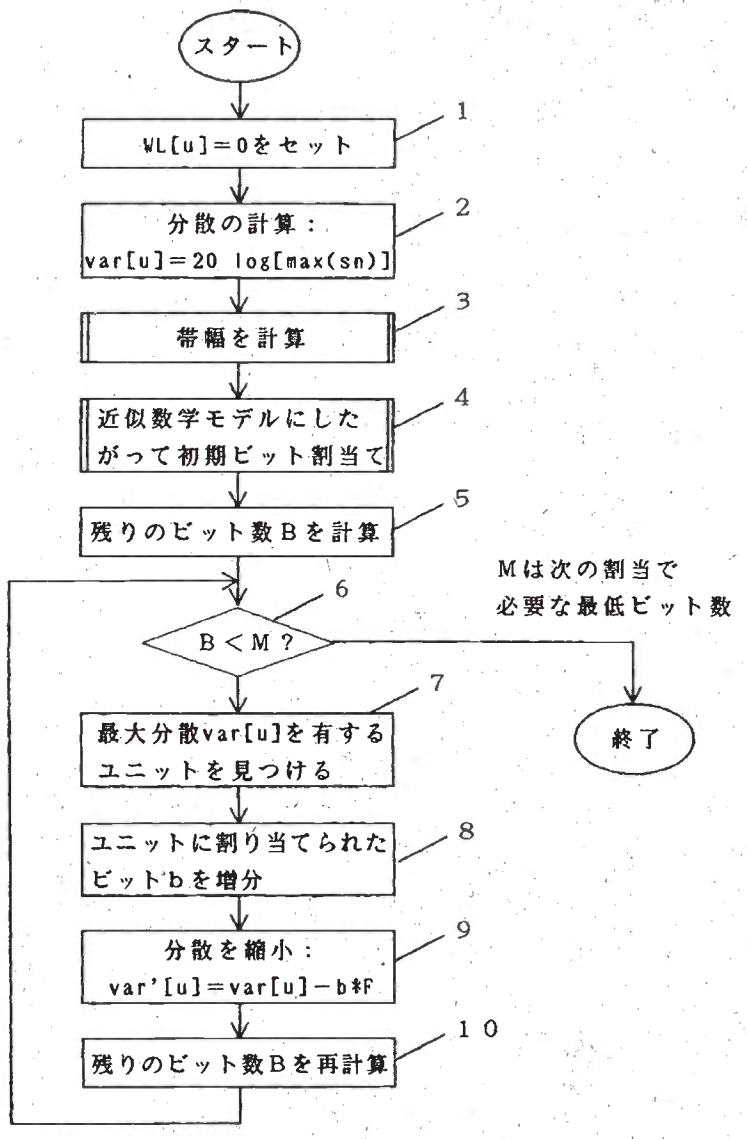
８：換算係数計算

９：動的ビット割当

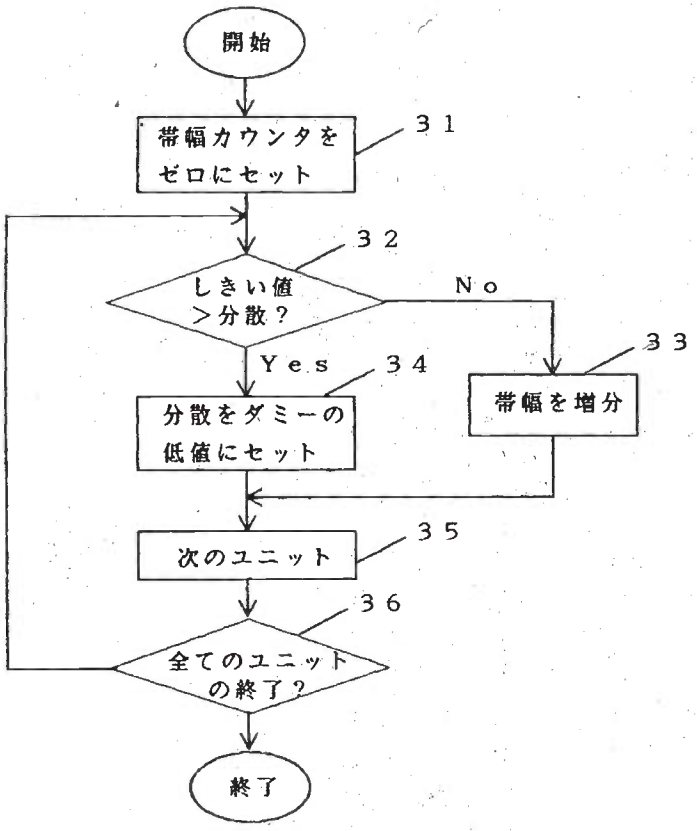
１０：量子化

【書類名】　図面

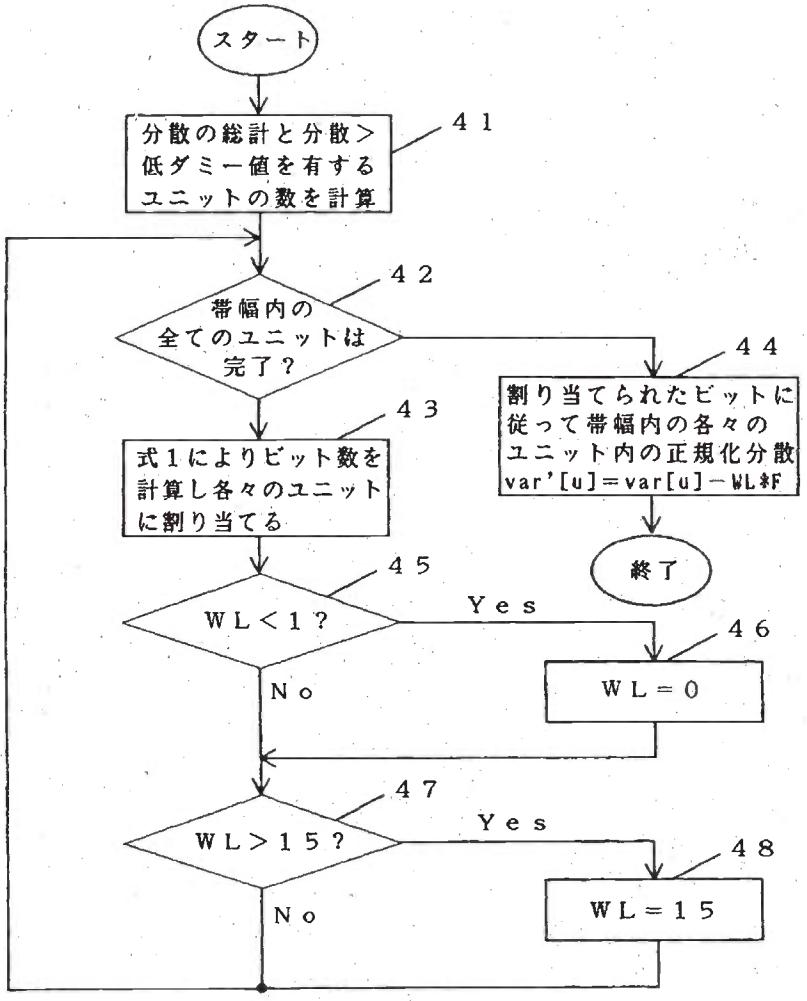
【図１】


スタート
WL[u]＝0をセット
1
分散の計算：
var[u]＝20 log[max(sn)]
2
帯幅を計算
3
近似数学モデルにしたがって初期ビット割当て
4
残りのビット数Ｂを計算
5
Mは次の割当で
必要な最低ビット数
6
Ｂ＜Ｍ？
終了
7
最大分散var[u]を有するユニットを見つける
ユニットに割り当てられたビットｂを増分
8
分散を縮小：
var'[u]＝var[u]－b*F
9
残りのビット数Ｂを再計算
１０

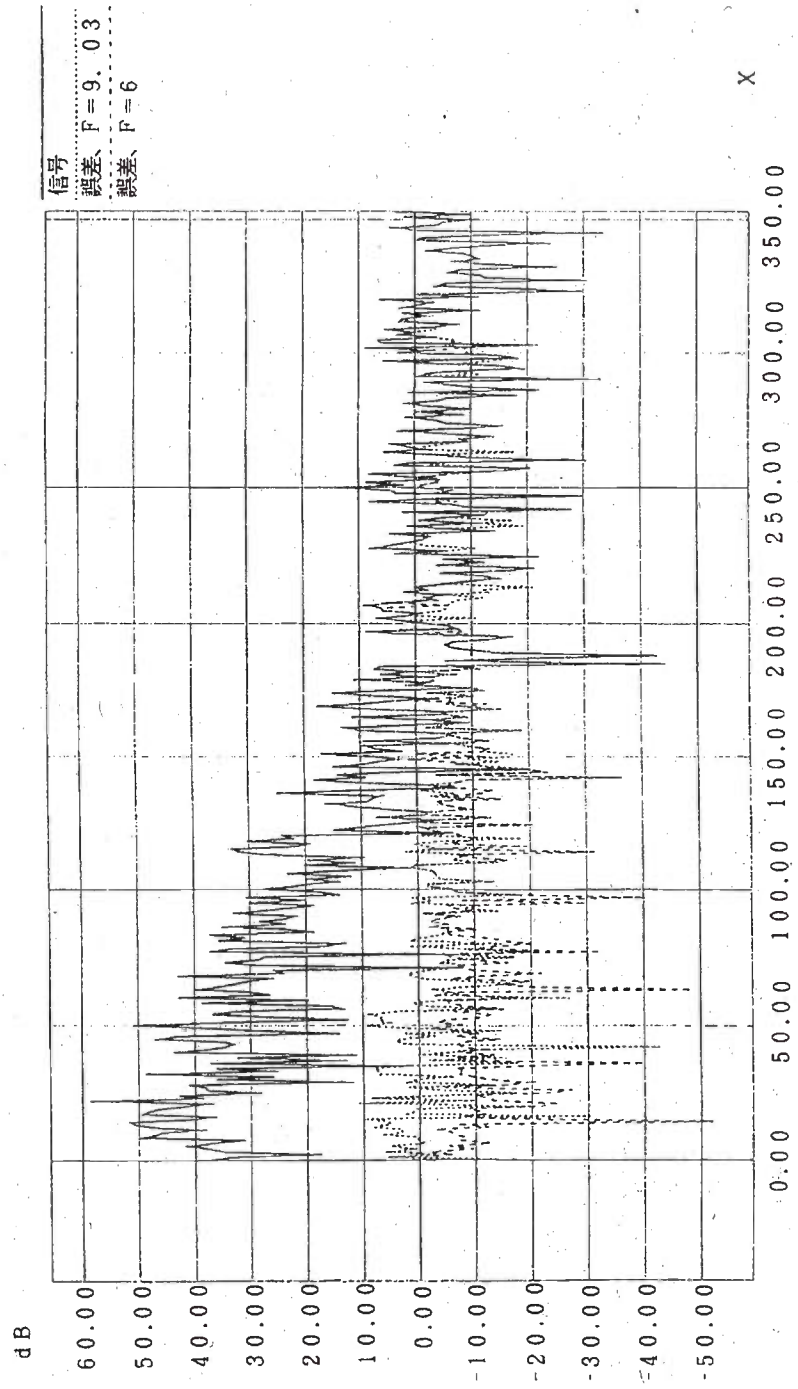
【図２】


開始
帯幅カウンタを
ゼロにセット
３１
３２
しきい値
＞分散？
Ｎｏ
Ｙｅｓ
３４
分散をダミーの
低値にセット
３３
帯幅を増分
次のユニット
３５
３６
全てのユニット
の終了？
終了

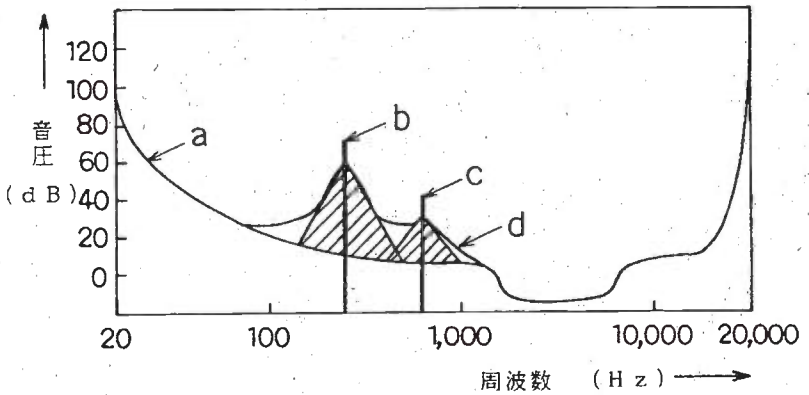
【図3】


スタート
分散の総計と分散＞低ダミー値を有するユニットの数を計算
41
帯幅内の全てのユニットは完了？
42
Yes
44
割り当てられたビットに従って帯幅内の各々のユニット内の正規化分散
var'[u]＝var[u]－WL*F
終了
No
43
式1によりビット数を計算し各々のユニットに割り当てる
45
WL＜1？
Yes
46
WL＝0
No
47
WL＞15？
Yes
48
WL＝15
No

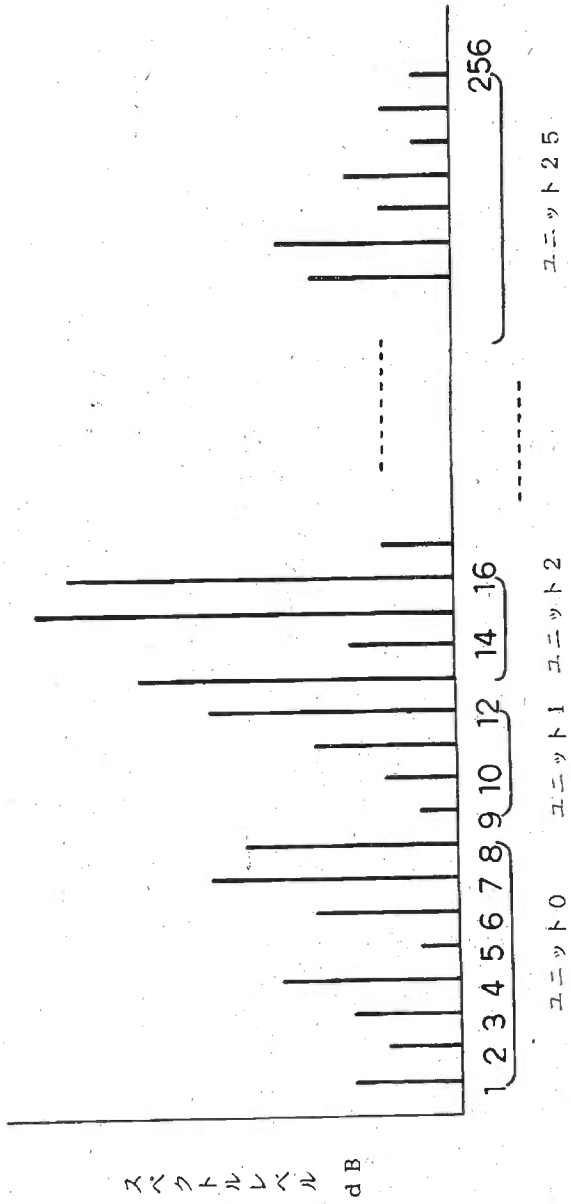
【図４】


信号
誤差、F＝9．03
誤差、F＝6
X
dB
60.00
50.00
40.00
30.00
20.00
10.00
0.00
-10.00
-20.00
-30.00
-40.00
-50.00
0.00
50.00
100.00
150.00
200.00
250.00
300.00
350.00

【図５】



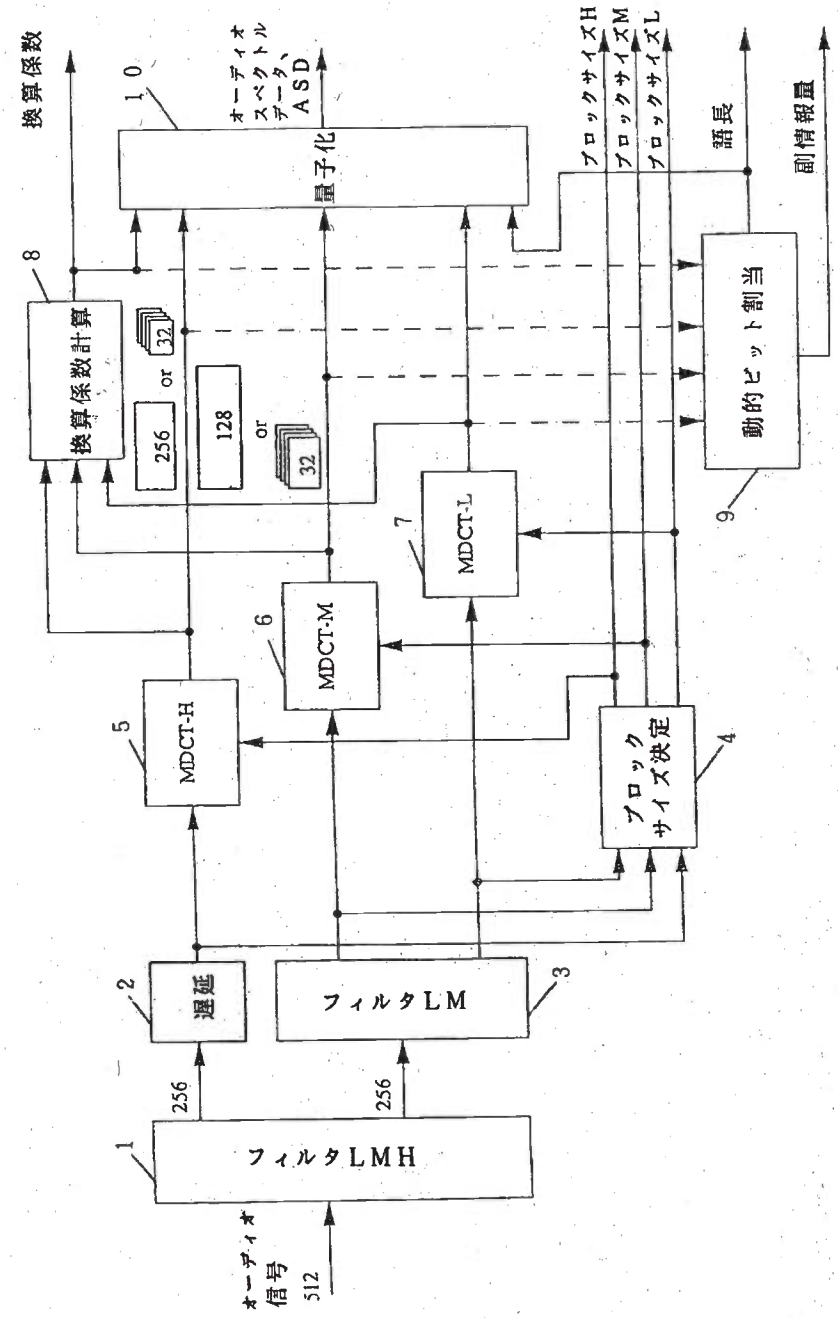
【図6】


スペクトルレベル
dB
1 2 3 4 5 6 7 8 9 10 12 14 16 256
ユニット0
ユニット1
ユニット2
ユニット25

【図７】

ユニット番号	スペクトル本数	ビット配分データ
ユニット０	８	ＷＬ（０）
ユニット１	４	ＷＬ（１）
ユニット２	４	ＷＬ（２）
ユニット３	４	ＷＬ（３）
ユニット２４	１０	ＷＬ（２４）
ユニット２５	１０	ＷＬ（２５）

【図８】


換算係数
オーディオスペクトルデータ、ASD
量子化
10
ブロックサイズH
ブロックサイズM
ブロックサイズL
語長
副情報量
8
換算係数計算
32
or
256
128
or
32
動的ビット割当
9
MDCT-L
7
MDCT-M
6
MDCT-H
5
ブロックサイズ決定
4
遅延
2
フィルタLM
3
256
256
フィルタLMH
1
オーディオ信号
512

【書類名】　要約書

【要約】

【目的】　低コスト目標を満たしつつ高オーディオ品質を達成できるような、簡単な構成の、人間の聴覚システムに適応するようにした動的ビット割当手順を設計すること。

【構成】　オーディオ信号がフレーム内でバッファされ、そのスペクトル成分に分解されるスペクトル及び時間的構造を有するディジタル・オーディオ信号の量子化を決定する際、所定の周波数間隔内の前記フレームの成分の分散及び／又は代表値を得て、心理音響学的基準を用いて必要な帯幅を計算し、すなわち無関係性を除去するため人間の聴覚の限界値を用いてオーディオ信号の必要な帯幅を判定し、信号の帯域内マスキング効果を考慮する粗いモデルを用いて前記周波数間隔の各々の初期量子化を決定し、モデルの効果を維持しつつ必要な最終的なビット速度を満たすため量子化を反復的に増加する。

【選択図】　図１

０５－０３０７１２

【書類名】　職権訂正データ
【訂正書類】　特許願

＜認定情報・付加情報＞
【特許出願人】
【識別番号】　000005821
【住所又は居所】　大阪府門真市大字門真１００６番地
【氏名又は名称】　松下電器産業株式会社
【代理人】　申請人
【識別番号】　100092794
【住所又は居所】　大阪府大阪市淀川区宮原５－１－３　新大阪生島ビル５Ｆ　松田特許事務所
【氏名又は名称】　松田　正道

出願人履歴情報

識別番号　〔０００００５８２１〕

１．変更年月日　１９９０年　８月２８日
［変更理由］　新規登録
住　所　大阪府門真市大字門真１００６番地
氏　名　松下電器産業株式会社

Transaction History Date 1995-10-02
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03-48683

PATENT

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicants: TEH, et al.

Serial No.: 08/161,798 Group Art Unit: 2104

Filed : December 6, 1993

For : A BIT ALLOCATION METHOD FOR TRANSFORM CODER

OCT 1 1995
GROUP 230

INFORMATION DISCLOSURE STATEMENT

Honorable Commissioner
of Patents and Trademarks
Washington, D.C. 20231

Sir:

Applicants respectfully submit a copy of the European Search Report, dated August 7, 1995, in their counterpart European patent application, together with a copy of each of the four references cited therein.

A PTO Form 1449 is attached hereto. The Examiner is respectfully requested to consider each of these references and to indicate such consideration on the attached Form 1449.

Respectfully submitted,

BEVERIDGE, DeGRANDI, WEILACHER & YOUNG

By: Joseph A. DeGrandi

Joseph A. DeGrandi, Reg. No. 17,446
Suite 800, 1850 M Street, N.W.
Washington, D.C. 20036
Telephone: (202) 659-2811

October 2, 1995

P.B.5818 - Patentlaan 2
2280 HV Rijswijk (ZH)
☎ (070) 3 40 20 40
TX 31651 epo nl
FAX (070) 3 40 30 16

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Grünecker, Kinkeldey,
Stockmair & Schwanhäusser
Anwaltssozietät
Maximilianstrasse 58
D-80538 München
ALLEMAGNE

EINGEGANGEN
3. AUG. 1995
Frist .....

Datum/Date
7. 08. 95

Zeichen/Ref./Réf.	Anmeldung Nr./Application No./Demande n°. //Patent Nr./Patent No./Brevet n°.
EP 9877-50/iw	94101199.1 - -

Anmelder/Applicant/Demandeur//Patentinhaber/Proprietor/Titulaire
MATSUSHITA ELECTRIC INDUSTRIAL CO., LTD.

COMMUNICATION

The European Patent Office herewith transmits

- [X] the European search report
- [ ] the declaration under Rule 45 EPC
- [ ] the partial European search report under Rule 45 EPC
- [ ] the supplementary European search report concerning the international application under Article 157(2) EPC relating to the above-mentioned European patent application. Copies of the documents cited in the search report are enclosed.

The following specifications given by the applicant have been approved by the Search Division :

- [X] Abstract
- [X] Title
- [X] Figure

- [ ] The abstract was modified by the Search Division and the definitive text is attached to this communication.
- [ ] The following figure will be published with the abstract, since the Search Division considers that it better characterises the invention than the one indicated by the applicant.

  Figure:

- [X] Additional copy(copies) of the documents cited in the European search report.

C. Lekahena

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REFUND OF THE SEARCH FEE

If applicable under Article 10 Rules relating to fees, a separate communication from the Receiving Section on the refund of the search fee will be sent later.

EPO Form 1507 02.93				

European Patent Office

EUROPEAN SEARCH REPORT

Application Number
EP 94 10 1199

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int.Cl.5)
A (1)	WO-A-89 07866 (QUEENS UNIVERSITY) * abstract * * page 1, line 23 - page 2, line 12; figure 13A * * figure 13B *	1	H04B1/66
A (2)	IEEE GLOBAL TELECOMMUNICATIONS CONFERENCE, vol.1, 2 December 1990, NEW YORK pages 518 - 522 MAHIEUX ET AL. 'TRANSFORM CODING OF AUDIO SIGNALS AT 64KBIT/S' * page 520, paragraph V * "THE BIT ALLOCATION"	1	
A (3)	PHILIPS JOURNAL OF RESEARCH, vol.44, no.2/3, July 1989, EINDHOVEN NL pages 329 - 343 VELDHUIS ET AL. 'SUBBAND CODING OF DIGITAL AUDIO SIGNALS' * page 338, line 3 - page 340, line 33 *	1	
A (4)	WO-A-92 17884 (SONY) * page 11, line 12 - line 16 * * page 22, line 18 - page 24, line 6 *	1	TECHNICAL FIELDS SEARCHED (Int.Cl.5) H04B

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
THE HAGUE	28 July 1995	Holper, G

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
Y : particularly relevant if combined with another document of the same category
A : technological background
O : non-written disclosure
P : intermediate document

T : theory or principle underlying the invention
E : earlier patent document, but published on, or after the filing date
D : document cited in the application
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ANNEX TO THE EUROPEAN SEARCH REPORT ON EUROPEAN PATENT APPLICATION NO. EP 94 10 1199

This annex lists the patent family members relating to the patent documents cited in the above-mentioned European search report. The members are as contained in the European Patent Office EDP file on
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28-07-1995

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
WO-A-8907866	24-08-89	AT-T- 112903	15-10-94
		DE-D- 68918846	17-11-94
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		EP-A- 0398973	28-11-90
		JP-T- 3504787	17-10-91
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		PT-A- 100309	29-04-94
		TR-A- 25961	01-11-93
		US-A- 5414795	09-05-95

EPO FORM P0459

For more details about this annex : see Official Journal of the European Patent Office, No. 12/82

INTERNATIONAL SEARCH REPORT

International Application No PCT/JP 92/00387

I. CLASSIFICATION OF SUBJECT MATTER (if several classification symbols apply, indicate all)*

According to International Patent Classification (IPC) or to both National Classification and IPC

Int.Cl. 5 G11B20/10; H04B1/66

II. FIELDS SEARCHED

Minimum Documentation Searched[7]

Classification System	Classification Symbols
Int.Cl. 5	G11B ; H04B

Documentation Searched other than Minimum Documentation to the Extent that such Documents are Included in the Fields Searched[8]

III. DOCUMENTS CONSIDERED TO BE RELEVANT[9]

Category *	Citation of Document,[11] with indication, where appropriate, of the relevant passages[12]	Relevant to Claim No.[13]
Y	EP,A,0 370 277 (SIEMENS AG) 30 May 1990 see the whole document	1-13,16, 17
Y	WO,A,9 009 064 (DOLBY LABORATORIES LICENSING CORPORATION) 9 August 1990 see abstract see page 7, line 18 - page 10, line 29 see page 22, line 38 - page 27, column 16; figures 1-26	3,7-13, 16,17
P,Y	EP,A,0 458 645 (SONY CORPORATION) 27 November 1991 see the whole document	4
Y	EP,A,0 206 352 (FUJITSU K.K.) 30 December 1986 see page 2, line 23 - page 3, line 29	1

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* Special categories of cited documents :[10]

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"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

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IV. CERTIFICATION

Date of the Actual Completion of the International Search	Date of Mailing of this International Search Report
23 JUNE 1992	14. 07. 92
International Searching Authority	**Signature of Authorized Officer**
EUROPEAN PATENT OFFICE	DEVERGRANNE C.

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III. DOCUMENTS CONSIDERED TO BE RELEVANT (CONTINUED FROM THE SECOND SHEET)

Category °	Citation of Document, with indication, where appropriate, of the relevant passages	Relevant to Claim No.
P,Y	EP,A,0 446 031 (SONY CORPORATION) 11 September 1991 see the whole document	2,5,6
A	EP,A,0 379 613 (DOLBY LABORATORIES LICENSING CORPORATION) 1 August 1990	
A	PROC. IEEE CONF. ON CIRCUITS AND COMPUTERS (ICCC82) 28 November 1982, NEW-YORK,US pages 474 - 477; BRANDENBURG ET AL: 'A DIGITAL SIGNAL PROCESSOR FOR REAL TIME ADAPTIVE TRANSFORM CODING OF AUDIO SIGNALS UP TO 20 KHZ BANDWIDTH'	

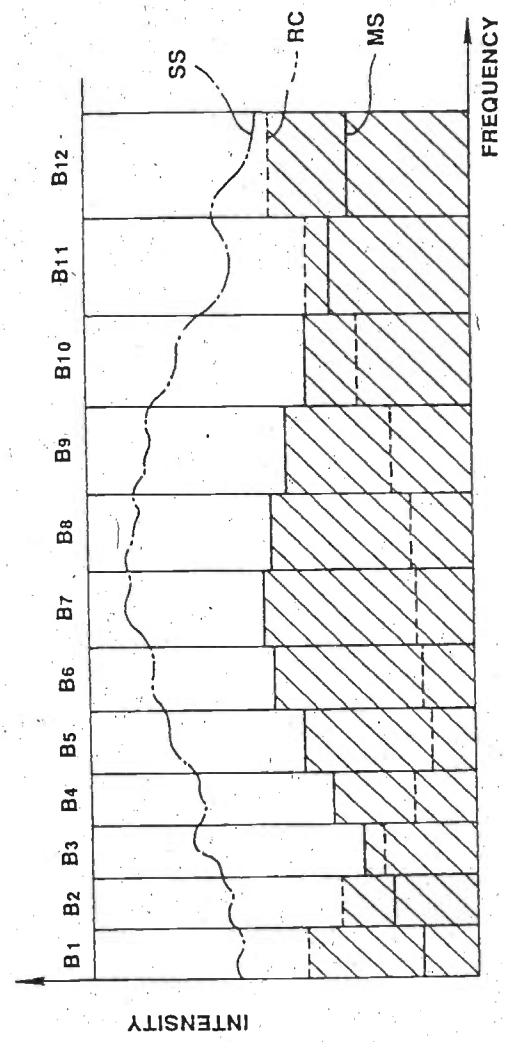

SS
RC
MS
FREQUENCY
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INTENSITY

FIG. 17

OCT 2 1995

Form PTO-1449 (Rev. 7-80) — U.S. Department of Commerce, Patent and Trademark Office

INFORMATION DISCLOSURE STATEMENT

(Use several sheets if necessary)

ATTY. DOCKET NO.	SERIAL NO.
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APPLICANT	
TEH, et al.	
FILING DATE	**GROUP**
December 6, 1993	~~2104~~ 2308

U.S. PATENT DOCUMENTS

*EXAMINER INITIAL		DOCUMENT NUMBER	DATE	NAME	CLASS	SUBCLASS	FILING DATE IF APPROPRIATE
	AA						
	AB						
	AC						
	AD						
	AE						
	AF						
	AG						
	AH						
	AI						
	AJ						
	AK						

FOREIGN PATENT DOCUMENTS

		DOCUMENT NUMBER	DATE	COUNTRY	CLASS	SUBCLASS	TRANSLATION YES	TRANSLATION NO
IC	AL	WO89/07866	8/24/89	PCT	—	—	X	
IC	AM	WO92/17884	10/15/92	PCT	—	—	X	
	AN							
	AO							
	AP							

OTHER INFORMATION (including author, title, date, pertinent pages, etc.)

IC	AQ	*IEEE Global Telecommunications Conference*, vol. 1, 12/2/90, New York, pp. 518-522, Mahieux, et al. "Transform Coding of Audio Signals at 64KBIT/S!," p. 520, ¶ V "The Bit Allocation."
IC	AR	*Philips Journal of Research*, vol. 44, no. 2/3, July 1989, Eindhoven NL, pp. 329-343, Veldhuis, et al. "Subband Coding of Digital Audio Signals," pp. 338, l. 3 - p. 340, l. 33.
	AS	
	AT	
	AU	

EXAMINER	DATE CONSIDERED
Indranil Chowdhury	6/9/96

* Examiner: Initial if reference considered, whether or not citation is in conformance with MPEP 609; Draw line through citation if not in conformance and not considered. include copy of this form with next communication to applicant.

H04B1/66 M

Pour le titre du livre
voir en seconde page

CH2827-2

XP 000218782

TRANSFORM CODING OF AUDIO SIGNALS AT 64 KBIT/S

Y. Mahieux and J. P. Petit

Centre National d'Etudes des Télécommunications
Route de Trégastel 22300 Lannion FRANCE.

p518-522

ABSTRACT

The coding of high quality sound in 64 kbit/s is of high interest for applications such as ISDN. The algorithm that we describe in this paper, allows the reduction at such a bit rate while keeping the original quality. It is based on transform coding and it uses a TDAC transformation. Perceptual properties and the interblock redundancy of the spectrum are involved when coding the transform coefficients. The complexity of this algorithm allows its real time implementation on one floating point digital signal processor such as the DSP 32C of ATT. Finally, the performances and the subjective results of this coding system are discussed.

DOC

I INTRODUCTION

The major drawback of the use of digital audio in Telecommunication applications such as ISDN, is the required bit rate. Many algorithms have been recently proposed to reduce this bit rate. Most of them are based on Transform Coding [1], [5]. They achieve a high quality coding at a bit rate of 2 to 3 bits per sample.

The coding system that we propose is designed to work at a bit rate of 64 kbit/s per monophonic channel for a sampling frequency, F_s, from 32 kHz to 48 kHz. But because of the flexibility of the bit allocation procedure, other bit rates are easily possible.

This algorithm looks like the well known Adaptive Transform Coding (ATC). Only the encoder is described in this paper. The structure of the decoder is quite symmetrical to the one of the encoder (figure 1).

After the TDAC transformation which is shortly described in section II, the masked coefficients are detected in the short term spectrum (section III). In order that encoder and decoder achieve the same bit allocation, a spectrum descriptor is computed from the non masked coefficients. Its coding scheme takes advantage of the long term stationarity of the signal (section IV). Then the section V describes the bit allocation procedure which is designed to perform the spectral noise shaping. The bits are distributed according to the perceptual importance of each coefficient. Finally, the section VI deals with the real time implementation and the performances of this algorithm.

II TRANSFORMATION AND WINDOWING

The transform that we use is the Time Domain Aliasing Cancellation (TDAC) transform, [2], which is defined as :

$$y(m,k) = \sum_{n=0}^{N-1} x(n).h(n)\cos\left(\frac{2\pi(2k+1)(2n+1)}{4N}+\frac{(2k+1)\pi}{4}\right) \quad (1)$$

$k = 0,\ldots,N-1$

-N is the block size, h(n) is the analysis window;
-m is the temporal index, k is the frequential index.

The inverse transform is :

$$x(m,n) = \sum_{k=0}^{N-1} y(m,k).f(n)\cos\left(\frac{2\pi(2k+1)(2n+1)}{4N}+\frac{(2k+1)\pi}{4}\right) \quad (2)$$

$n = 0,\ldots,N-1.$

- f(n) is the synthesis window.

The reconstructed signal is simply obtained by overlap-add of x(m,n). The perfect reconstruction is achieved with the windows h(n) = f(n) defined as :

$$h(n) = \sqrt{2}.\sin(\pi.(n+1/2)/N) \quad (3)$$

The computation of this transform is performed by a particularly efficient algorithm. The global amount of required operations is about $N.\log_2(N)$. The coding algorithm has been designed with a N=1024 sample window which delivers a relatively high spectral resolution. Because of the limitation of the human ear bandwith, only the N_t first coefficients are coded. N_t corresponds to the frequency of 20 kHz. (see Table 1).

III DETECTION OF THE MASKED COEFFICIENTS.

Because of frequential masking [3], some of the spectral lines y(m,k) are inaudible. They may be non transmitted without adding any perceptible degradation to the original signal. At the encoder, they are detected by comparing the energies $y^2(m,k)$, to the masking threshold S(m,k). Then the masked coefficient indexes are transmitted to the decoder by "Run Length Coding".

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1) Computation of the masking threshold

a) Principles of the computation

The computation of the masking threshold is performed on a line by line basis. First the spectrum is multiplied by the transmission factor of the ear, $a_0(k)$ [3]. This weighting is equivalent to lower the energy of the highest frequencies. Then the masking threshold is derived from the convolution of the spectrum $Y(\nu)$ expressed on the critical band rate (also called Bark scale), ν, by the basilar membrane spreading function $B(\nu)$. The influence of each transform coefficient $y(m,k)$ to the masking threshold of each other coefficient $y(m,i)$, $i \neq k$ can be expressed as :

$$S_k(m,i) = B(\nu_i - \nu_k).a_0(k).y^2(m,k) \qquad (4)$$

The global masking threshold at the spectral line $y(m,i)$ is :

$$S(m,i) = \theta.\sum_{k=0}^{N_t-1} B(\nu_i - \nu_k).a_0(k)y^2(m,k) \qquad (5)$$

The threshold $\theta = 0.001$ corresponds to the signal to noise ratio for the complete masking of a tone by another tone located into the same critical band [3]. The basilar membrane spreading function $B(\nu)$ and therefore the computation of equation (5) can be splitted into 3 parts. The first one, named central masking, refers to the values of ν close to 0 where $B(\nu)$ can be roughly approximated as a constant :

$$B(\nu) = 1 \quad -1/2 \text{ Bark} < \nu \leq 1/2 \text{ Bark} \qquad (6)$$

The two other ones, called side masking, are the slopes of $B(\nu)$, that is the masking towards higher and lower critical bands (i.e. for $|\nu| > 1/2$ Bark). There, $B(\nu)$ is estimated as a straight line whose slope is -10 dB per critical band towards higher frequencies and -27 dB per critical band towards lower frequencies (see the figure 2).

b) Fast computation of S(m,k)

The TDAC transform delivers a regularly frequency spaced spectrum, whereas $B(\nu)$ refers to the Bark scale which is a non linear function of the frequency. To solve this problem, the Bark scale is assumed to be constant over a low number of neighbouring coefficients.

On the other hand, a direct evaluation of the equation (5) would require a too high amount of computations. However, due to the peculiar shape of $B(\nu)$, it can be drastically reduced. For instance, for each coefficient $y(m,k)$, the side masking, $S_{\alpha}(m,k)$ due to the lower frequency coefficients, can be partly derived from the previous one, $S_{\alpha}(m,k-1)$. In the same way, $S_{\beta}(m,k)$ (side masking resulting from higher frequency coefficients) can be computed from $S_{\beta}(m,k+1)$. Actually, the 3 parts of the computation of $S(m,k)$ are rewritten to a recursive form which is much more efficient regarding the computational load. Finally, the coefficient $y(m,k)$, is masked if :

$$a_0(k).y^2(m,k) < S(m,k) \qquad (7)$$

The number of masked coefficients per transform block, is very variable. It is especially high for broadband signals. For instance, the average number of masked coefficients is 395 over 512 for a sequence of bassoon of duration 3 seconds.

2) Run Length Coding

a) Coding principle

Let $I(m,k)$ be a bit equal to 1 if the coefficient $y(m,k)$ is masked. The series $\{I(m,k), k=0,...,N_t/2-1\}$ exhibits more or less long unbroken runs of 0 or of 1. The length of each run is coded and transmitted, so allowing the decoder to rebuild the whole range $\{I(m,k), k=0,...,N_t/2 - 1\}$. Run Length Coding is efficient when associated with entropy coding [4]. Optimised Huffman codes have been computed in each case (runs of 0 and runs of 1).

The Run Length Coding starts after the first masked spectral line whose index is coded on 9 bits. At the decoder, the Huffman codes are decoded until the upper bound of the spectrum is reached. Then, the masked coefficients are simply forced to zero

b) Run Length Coding restriction

To make sure that the non transmission of masked coefficients yields a positive gain of bits, the number of used bits, $R_d(m)$ (i.e. the sum of the Huffman codeword lengths) is compared to the number of saved bits, $R_s(m)$. $R_s(m)$ is the total number of the bits that would be allocated to the masked spectral lines if they were not forced to zero. For each transform coefficient this number is estimated by an average value previously computed. If the gain $G(m) = R_s(m) - R_d(m)$ is negative, then this comparison is locally performed. At each run of masked coefficients, a local value $G(m,k)$ is computed and the non transmission of the masked spectral lines is confined to the frequency regions where this gain is positive.

By using this procedure, the masking of coefficients is particularly attractive. The average ancillary bit rate is about 0.8 bit per masked coefficient.

IV SPECTRUM DESCRIPTOR CODING

1) The Spectrum Descriptor computation

As shown in figure 1, a spectrum descriptor is transmitted at each block enabling the encoder and the decoder to compute the same bit allocation. For an efficient bit allocation procedure, this spectrum descriptor must be as precise as possible while requiring a low ancillary bit rate.

The spectrum descriptor is defined as the standard deviation $\sigma(m,j)$ of the non masked coefficients in unequal width frequency bands. The number of bands, N_{sd}, depends on the sampling frequency (see Table 1).

2) Coding of the spectrum descriptor

The set $\{\sigma(m,j)\}$ is largely uncorrelated. Coding techniques such as vector quantization are inefficient. But, the spectrum descriptor is highly time correlated. Indeed, the frequential distribution of the energy is slowly varying from one block to another. In case of transients, this temporal correlation vanishes. Therefore, the coding of the spectrum descriptor is performed by one coding scheme chosen among 2 different strategies.

a) Temporal Predictive Coding

The most direct mean to take advantage of the inter-block redundancy, is to perform a predictive coding which is applied to the spectrum descriptor expressed in dB. This predictive coding scheme is made up of a convertor to log values, a DPCM encoder and a variable length encoding system (figure 3). The predictor is a first order one whose prediction coefficient a_1 is very close to 1. A higher prediction order does not give a significant improvement.

The prediction error is :

$$e(m,j) = \log(\sigma(m,j)) - a_1.\log(\sigma'(m-1,j)) \qquad (8)$$

$$j = 0,..., N_{sd} - 1.$$

- $\sigma'(m,j)$ is the quantized version of $\sigma(m,j)$.

$e(m,j)$ is quantized by means of a 16 level uniform quantizer whose step size Δ equals 5 dB. Because of the stationarity of the signal, the codeword density of probability is highly concentrated. By using an entropy coding of the quantizer outputs, less than 2.0 bits are sufficient instead of 4 bits.

b) Frequential Predictive Coding

For this second quantization mode, a frequential predictive coding is performed. The prediction error is :

$$e(m,j) = \log(\sigma(m,j)) - \log(\sigma'(m,j-1)) \qquad (9)$$

$$j = 1,..., N_{sd} - 1.$$

$e(m,j)$ is coded by means of a 16 level uniform quantizer also combined with a Huffman encoder. The first value, $\sigma(m,0)$ is "log. quantized" by a 7 bit quantizer.

The spectrum descriptor coding mode is chosen according to the required amount of bits. The two predictive coding schemes are successively performed. The one which requires the lower bit rate is retained. To indicate this choice, one bit is transmitted to the decoder.

V THE BIT ALLOCATION

1) General description

The bit allocation procedure distributes the bit rate between the spectral components, in such a way that the quantization noise is masked. It is designed to minimize the ratio Coding Noise/Masking Threshold, i.e. for the block m

$$X(m) = \sum_{k=0}^{N_t/2-1} \sigma_q^2(m,k)/S(m,k) \qquad (10)$$

- $\sigma_q^2(m,k)$ is the variance of the quantization noise for the coefficient $y(m,k)$.
- $S(m,k)$ is the masking threshold.

$X(m)$ is minimized by including the constraint $\sum R(m,k) = R_0(m)$. $R(m,k)$ is the number of bits allocated to $y(m,k)$ and $R_0(m)$ is the number of bits to distribute. This minimization yields the following equation :

$$R(m,k) = 1/2.\log_2(\alpha(m,k).\epsilon.a_0(k).\sigma^2(m,k)/S(m,k)) + \lambda \qquad (11)$$

where λ is a constant, $a_0(k)$ is the transmission factor of the ear. ϵ is the performance factor of the quantizer [4]. $\sigma^2(m,k)$ is the variance of the coefficient $y(m,k)$. The additional factor $\alpha(m,k)$, allows to involve the flatness (or the peakness) of the spectrum [5].

At the decoder, only the quantized values of the spectrum descriptor are available. Then the number of bits is computed at each frequency group j from $\{\sigma'(m,j)\}$. All the non masked coefficients $y(m,k)$ in the same band j receive the same number of bits. The masking threshold $S(m,k)$, previously computed for the detection of the masked coefficients is unavailable at the decoder. A new masking threshold $S_1(m,j)$ is derived at the encoder and at the decoder from $\{\sigma'(m,j)\}$. Its computation is similar to the one described in section III-1.

Therefore : $R(m,j) = 0.5.\Psi(m,j) + \lambda \qquad (12)$

where

$$\Psi(m,j) = \log_2(\alpha(m,j).\epsilon.a_0(j).\sigma'^2(m,j)/S_1(m,j) \qquad (13)$$

$$j = 0,..., N_{sd} - 1.$$

2) Computation of $\Psi(m,j)$

The number of non zero coefficients in each frequency band is different (due to the non equal width of the bands and to the detection of masked coefficients). Then the performances of the quantizer are different too. Actually, 4 sets of quantizers have been computed (from 1 to 7 bits), each one corresponding to a different number of non zero coefficients in the frequency band.

The constant $\alpha(m,j)$ is derived from the local spectral flatness measures ("sfm"), $\{\gamma(\ell), \ell = 1,..,4\}$, which are computed in 4 frequency bands of equal width [4]. The 4 sfm's values are encoded by means of a 6 bit uniform quantizer whose range is [-64 dB, 0 dB]. The function $\alpha(m,j)$ contributes to raise the perceptual importance of the peaks. If the spectrum is "sharper", the masking threshold has to be lowered [5]. So, for a better subjective quality, the peaks of the spectrum must receive a higher number of bits.

3) Bit Allocation procedure

By iterating the value of λ, the equation (12) is solved

so that {R(m,j)} are integer values which verify the constraint. The initial value of λ is given by :

$$\lambda_0(m) = (0.5.\sum_{k=0}^{N_t-1} \Psi(m,k) - R_0(m))/(N_t/2 - N_{mc}) \quad (14)$$

N_{mc} is the total number of masked coefficients.

Then, R(m,j) is derived by

$$R(m,j) = I\{0.5.\Psi(m,j) + \lambda\} \quad (15)$$

where $R_{max} \geq R(m,k) \geq 0$ and I{.} means "nearest integer".

If $\sum R(m,k)$ is close to $R_0(m)$, or if the number of iterations exceeds a certain value (e.g. 16), the procedure is stopped. Otherwise, $\lambda_i(m)$ is modified :

$$\lambda_{i+1}(m) = \lambda_i(m) + (\sum R(m,k) - R_0(m))/(F_{cs}.N_\bullet^i) \quad (16)$$

$N_\bullet^i$ is the number of coefficients y(m,k) which received a positive number of bits at the step i and F_{cs} is the number of times that the sign of $\sum R(m,k) - R_0(m)$ has been changing. So, when $\sum R(m,k)$ is close to $R_0(m)$, the variation of $\lambda_i(m)$ is gradually reduced. Then the procedure is reset to step 2.

Generally, less than 3 iterations are sufficient to reach the convergence state. If at the end of this iterating procedure, $\sum R(m,k) \neq R_0(m)$, bits are added or subtracted according to an algorithm similar to the one described in [6].

VI IMPLEMENTATION AND RESULTS

1) Transmission error sensitivity

Although this algorithm is mainly devoted to ISDN applications with very low bit error rates, some precautions must be taken to avoid, for instance, the infinite propagation of errors. The sensitivity of the coding scheme to transmission errors is increased by the use of inter block prediction of the spectrum descriptor and of Huffman coding which generates a variable repartition of the bit rate inside the transform block.

To reduce this sensitivity, the bit stream may be rearranged and some ancillary informations have to be transmitted. For instance, the numbers of bits required by the Huffman codes (Run Length Coding and spectrum descriptor encoding) are transmitted at the beginning of each block. This allows the decoder to start the decoding of the remainder of the stream at the right place, even if a transmission error occurs.

Likewise, the temporal predictive coding of the bit stream is periodically reinitialised by using the frequential coding mode, even if the signal is stationary. So, the infinite propagation of transmission errors effects is avoided.

2) Complexity of the algorithm

The encoding algorithm requires about 12000 products and 18000 adds per block. Some particular mathematical operations have also to be performed (80 divisions per block and 120 logarithms). At the decoder, the number of operations is lower.

This computationnal load enables the real time implementation of the presented coding scheme. A floating point digital signal processor has been chosen in order to avoid the problems of precision which would be very critical with a fixed point format mainly because of the size of the transformation. The algorithm is now being implemented on a DSP 32C of ATT. At a sampling frequency of 32 kHz, one DSP 32C is sufficient to implement the encoder. The memory requirements are about 10 kBytes ROM and 12 kBytes RAM for the data aera.

This algorithm delivers a high quality sound at 64 kbits/s. For most pieces of music, it is very difficult to discriminate the original and the coded version. Subjective tests indicated that the algorithm works very well in case of broadband signals with many harmonics. For signals with flatter spectrum, the detection of masked coefficients is less interesting (or even does not work), yielding therefore a lower quality. This is particularly true for some critical pieces of song.

CONCLUSION

Table 1 sums up the main characteristics of this coding algorithm and Table 2 presents a typical repartition of the bit rate inside one transform block. It is clear that the detection of masked coefficients and the predictive coding of the spectrum descriptor yields a high average number of available bits per non masked coefficient.

This algorithm is a good compromise between a high quality coding at 64 kbit/s and a limited complexity which allows its real time implementation. Finally, the coding scheme described in this paper is well suited for ISDN applications such as the transmission of a stereo high quality sound in a "2B" channel environment.

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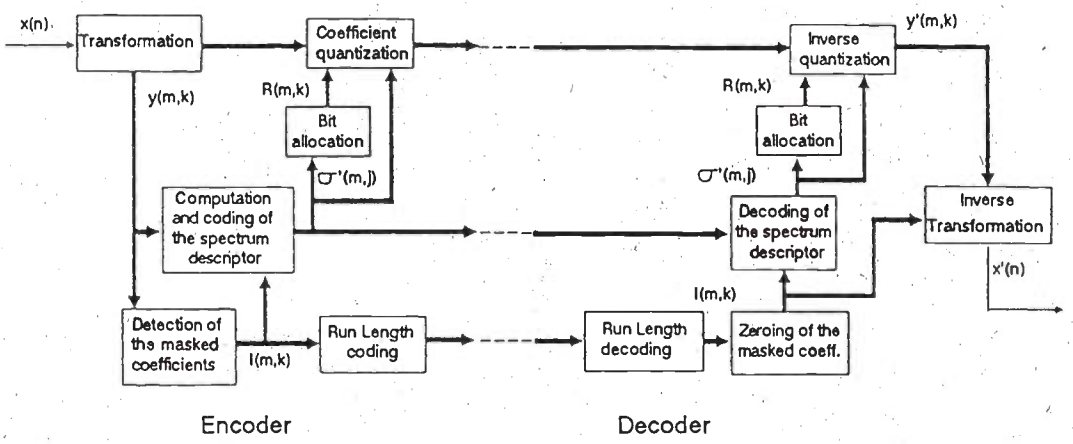


Figure 1 General Scheme of the Algorithm

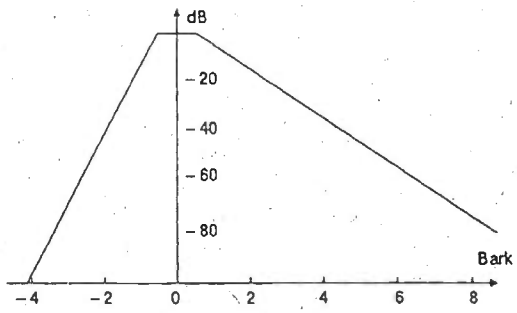


Figure 2 Basilar membrane spreading function

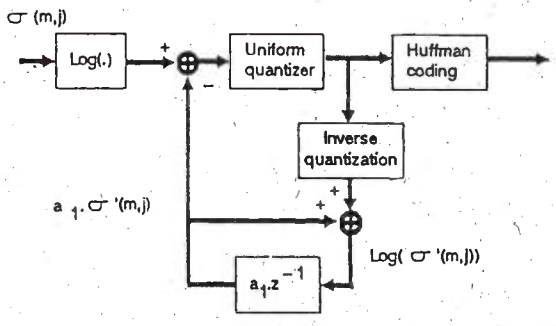


Figure 3 Predictive coding of the spectrum descriptor

Size of the Transform	N = 1024
Maximal number of coded coefficients sampling frequency F = 32 kHz F = 48 kHz	N_t = N/2 = 512 N_t = 427
Number of available bits per block F = 32 kHz F = 48 kHz	1024 bits 682 bits
Number of frequency groups for the spectrum descriptor F = 32 kHz F = 48 kHz	N_{sd} = 50 N_{sd} = 35

Table 1 Main characteristics of the algorithm

Ancillary information (Number of bits for Run Length Coding	21 bits
The 4 local sfm' s values	24 bits
Number of masked coefficients	378 coefficients
Run Length Coding	343 bits
Coding of the spectrum descriptor	121 bits
Bits to allocate to the non masked coefficients	515 bits

Table 2 Example of a bit stream repartition
(sampling frequency : 32 kHz).

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SUBBAND CODING OF DIGITAL AUDIO SIGNALS

by R.N.J. VELDHUIS, M. BREEUWER and R.G. VAN DER WAAL

Philips Research Laboratories, P.O. Box 80000, 5600 JA Eindhoven, The Netherlands

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Abstract

A subband coding method for high-quality digital audio signals is described. To achieve low bit rates at a high quality level, it exploits the simultaneous masking effect of the human ear. It is shown how this effect can be used in an adaptive bit allocation scheme. The method is capable of reducing the bit rate of a compact disk signal by a factor of seven. Results obtained with a low-complexity and a high-complexity system are discussed.

Keywords: adaptive bit allocation, digital audio, psycho-acoustics, simultaneous masking, source coding, subband coding

1. Introduction

Transmission and storage of high-quality digital audio signals are becoming important for the audio industry, for instance in the case of digital radio and of new applications for optical disks. The bit rate of a high-quality stereophonic digital audio signal, as it is recorded on a compact disk, is about 1.4 Mbit s^{-1}. For some transmission channels or storage media this is too high and therefore source coding is required. Since digital audio is associated with high quality, a perceptible loss of quality cannot be tolerated.

Source coding of audio signals at low bit rates generally introduces errors. This paper describes a coding method that attempts to keep coding errors inaudible by exploiting the simultaneous masking effect occurring in the human auditory system. This is the perceptive phenomenon that a weak signal, e.g. quantization noise, is masked (i.e. made inaudible) by a stronger signal, e.g. a pure tone in the audio signal. Simultaneous masking is briefly explained in Sec. 2.

Simultaneous masking is most effective if both masked and masking signal are in a rather narrow frequency band. This suggests the use of subband

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coding, where the signal is first split up into frequency bands which are then quantized. The structure of the subband coding system is given in Sec. 3.

Quantization should be such that the quantization noise is masked by the audio signal. This is achieved by using uniform block companded quantization [1]. The subband signals are split up into blocks. Each block is scaled to a unit level and then quantized by a uniform quantizer. Quantized data and scale factors are transmitted. In this manner the power of the quantization noise can be controlled by allocating a certain amount of bits to each quantizer.

In a subband coding system we can distinguish in-band masking, where both masked and masking signal are in the same subband, and out-of-band masking, where masking and masked signal are in different subbands. Both are exploited in the method described in this paper. Section 4 explains how the maximum power of the quantization noise that is masked, called the masked power, can be estimated for each subband.

Once the masked powers have been computed for all subbands, bits are allocated to the quantizers. Ideally the number of bits for each quantizer should be such that in each band the quantization noise is completely masked. However, the masked powers are signal dependent and therefore the number of bits needed to ensure complete masking varies in time. Here coding systems with a fixed bit rate are considered. Therefore the available bits must be divided over the subbands in such a way that the audible degradation of the output signal is minimal. This requires an adaptive bit allocation method, which is described in Sec. 5.

There is a trade-off between quality, bit rate and complexity. Complexity is largely determined by the splitting and merging subband filters. It can be kept low by keeping the number of subbands low and their minimum bandwidth high. At a fixed quality level, the lowest bit rate achievable with a 'low-complexity' system is higher than with more complex systems with more and narrower subbands. This is explained in Sec. 4. With a complex system a reduction in bit rate by a factor of seven can be obtained without a perceptible loss in quality. Results obtained with a simple and a complex system are discussed in Sec. 6. Section 7 summarizes the main results and gives some topics for future research.

2. Simultaneous masking

Simultaneous masking is the effect that a weak signal is made inaudible by a simultaneously occurring stronger signal. Masking has been discussed in great detail in refs 2 and 3. The use of masking in subband coding has been

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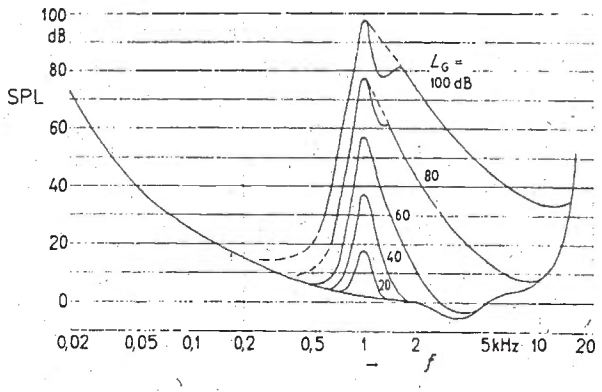


Fig. 1. Hearing threshold (lower curve) and masking thresholds of narrow-band noise signals centred at 1.0 kHz at various SPLs L_G.

described in refs 4 and 5. In the next two paragraphs results from ref. 2 are repeated to illustrate the masking effect.

Consider as a test signal one pure tone. This is inaudible if its sound pressure level (SPL)*) is below the threshold of hearing. The threshold of hearing is a function of frequency, as is shown in fig. 1*). In the presence of a second, stronger signal, the SPL above which the test signal is audible differs from the threshold of hearing. It is raised for frequencies close to the frequency of the stronger signal. This new threshold is called the masking threshold and the second, stronger signal is called the masking signal. The test signal is masked if its SPL is below the masking threshold. This is illustrated in fig. 1, where masking thresholds of narrow-band noise signals with a bandwidth of 90 Hz, centred at 1 kHz, at various sound pressure levels L_G are shown.

The masking threshold, as a function of frequency, depends on the SPL and the spectrum of the masking signal. For instance, masking thresholds for a pure tone as a masking signal are depicted in fig. 2. Figure 3 shows masking thresholds of narrow-band noise signals with different centre frequencies f_m. Masking thresholds of narrow-band noise signals have different shapes for centre frequencies below and above 500 Hz, as is illustrated by fig. 3. Furthermore, the masking thresholds of pure tones are lower and of a different shape than those of narrow-band noise signals, as illustrated by figs 1 and 2.

*) Figures 1, 2, and 3 are taken from ref. 2.

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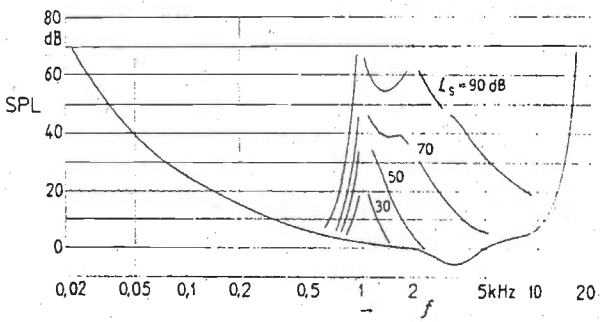


Fig. 2. Masking thresholds of pure tones of 1.0 kHz at various SPLs L_S.

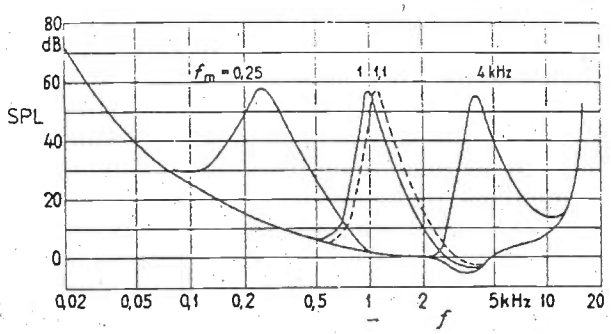


Fig. 3. Masking thresholds of narrow-band noise signals centred at various frequencies f_m.

The masking thresholds for both pure tones as well as for narrow-band noise signals show an asymmetry around the frequency of the masking signal. Signals with frequencies higher than the frequency of the masking signal are better masked than signals with frequencies lower than the frequency of the masking signal.

So far, masking of a pure tone by another pure tone or by a narrow-band noise signal has been considered to illustrate the masking effect. What is needed is some kind of simple, analytical masking model that can be used in a source coding system. This will be introduced in Sec. 4.

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The results presented in this section have been obtained from experiments performed on a large group of people [2,3]). They are valid for young people with good hearing. Older people will generally have a higher threshold of hearing at the higher frequencies, say above 10 kHz.

3. Subband coding

It is clear from fig. 1 that masking is strongest for frequencies close to the frequency of the masking signal. This suggests that the masking phenomenon can be well exploited in a subband coding system. In such a system the encoder splits up the signal into frequency bands, called subbands, which are then quantized. The quantized subband samples, together with some additional data, are transmitted. The decoder reconstructs the subbands from the received quantized samples and the additional data. Then the subbands are merged into a replica of the original signal.

The splitting of the signal into subbands and the merging of the subbands into a replica of the original signal are carried out by decimating and interpolating filter banks, such as quadrature-mirror or conjugate quadrature-mirror filter banks [7,8]). Because of the decimation, the sampling frequency of a subband signal equals twice the subband's bandwidth. Therefore the total sample rate after splitting is the same as the sample rate at the input. The ratio of the sampling frequency of the input signal of the filter bank and the sampling frequency of a signal in a subband is called the decimation factor of that subband. Because, as can be seen from fig. 3, the 'bandwidth' of the masking threshold increases with the frequency of the masking signal, the bandwidths of the subbands may increase with frequency.

In the process of quantization, quantization noise is added to the signals. If the filter banks have good frequency-separating properties, the quantization noise remains in the subband it was added to. It is masked by the audio signal if the signal-to-noise ratio is above a certain threshold. This implies that the quantizers must operate at a predetermined signal-to-noise ratio. This can be achieved with uniform block companded*) quantizers [1]). In this type of quantizer the signal is first divided into blocks. Of these blocks the maximum absolute values, called peak values, are computed. By dividing the samples in the blocks by the peak values, they are scaled to a unit level. The scaled blocks are then quantized with a uniform quantizer. The signal-to-noise ratio expressed in dB is proportional to the number of bits used in the quantizer, so that the signal-to-noise ratio of a quantizer can be predetermined by allocating a certain amount of bits to it.

*) To 'compand' is a combination of the verbs to compress and to expand.

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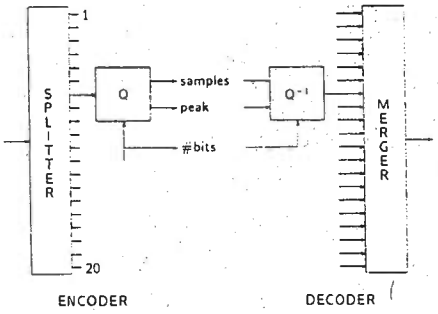


Fig. 4. Subband coding system with 20 subbands.

Figure 4 shows a diagram of a coding system with 20 subbands. As can be seen, quantized samples as well as coded peak values and side information to indicate the number of bits used for quantization are transmitted.

The numbers of samples in the blocks should be chosen sufficiently small so that changing statistics in the audio signal can be tracked. Otherwise it can no longer be assumed that the signal-to-noise ratio of the quantizer is constant. However, the number of samples in a block must not be too small, because then the bit rate increases as a result of the increased amount of side information. A maximum block length corresponding to 10–20 ms turns out to give good results. For instance, for the 20-band system of fig. 4 blocks of length 32, corresponding to 20 ms, have been chosen.

In Sec. 4 it is shown how for a given division of the signal into subbands the simultaneous masking effect described in Sec. 2 can be used to determine the masked powers in the subbands. How the final bit rate depends on the division into subbands is also explained. Section 5 shows how the number of bits allocated to each quantizer is computed.

4. A masking model for subband coding

In Sec. 2 results from psycho-acoustical literature have been presented to illustrate the simultaneous masking effect. These results cannot be used directly to compute masked powers in subbands as is needed in subband coding systems. For instance, in Sec. 2 masking of one pure tone by one other pure tone or by a narrow-band noise signal has been discussed. Music is a time-varying complex of tones combined with other noisy signals. Although

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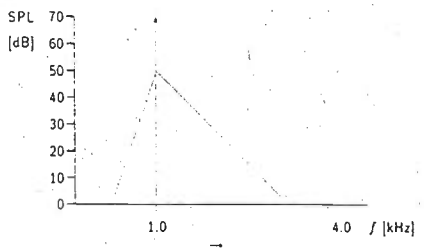


Fig. 5. Masking threshold as a function of frequency.

some experiments have been performed in that direction [9,10]), it is not known how the masking threshold of a complex of tones must be computed. Therefore it is unclear how, given a short-time music spectrum, masked powers in subbands must be computed.

As has been done in other audio source coding systems [5,11]) a masking model has to be assumed that provides answers to the above problems. It must be emphasized that this model is partly based on assumptions that have never been completely justified by experiments. The masking model chosen here is simple. More elaborate models will lead to better coding systems but with an increased complexity. As will be demonstrated in Sec. 6, good enough results can already be obtained with the simple model used here.

As a basis for the masking model the masking threshold of a pure tone is used. It is in this model assumed that, on a logarithmic scale, its shape is independent of the frequency of the tone and its SPL. This implies that for frequencies below 500 Hz the computed masking threshold will be too low. In the model used here the shape of the masking threshold of a pure tone is approximated by

$$T(f_m, f) = \begin{cases} T_{max}(f_m)\left(\dfrac{f}{f_m}\right)^{28}, & f \leqslant f_m, \\ T_{max}(f_m)\left(\dfrac{f}{f_m}\right)^{-10}, & f > f_m. \end{cases}$$

In this expression f_m is the frequency of the masking signal and $T_{max}(f_m)$ is the relative masking threshold at this frequency. $T_{max}(f_m)$ depends on the frequency of the masking signal [4]). The masking threshold of a tone is obtained by multiplying $T(f_m, f)$ with the power of the tone. Figure 5 shows a stylistic approximation of the masking threshold according to the masking model of

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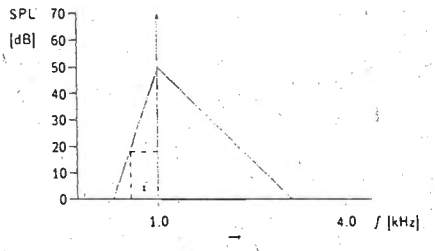


Fig. 6. In-band masking.

this paper for a pure tone of 1000 Hz at an SPL of 70 dB. For a frequency of 1000 Hz, $T_{max}(f_m)$ equals -20 dB [2]). The model ignores the effects occurring around the frequency of the masking tone and its harmonics that can be observed in fig. 2, but it describes the masking threshold with sufficient accuracy.

Furthermore, it is assumed that masking is additive: the masking threshold for a signal containing more than one frequency component can be obtained by adding the masking thresholds of the components. That this is allowed can be concluded from results in ref. 9, although some of these results have not been confirmed in ref. 10. From an engineering point of view the assumption of additive masking is very attractive. Otherwise a straightforward computation of masked powers would not be possible.

The masking model used here is a simplification of reality. Coding systems based on it may show unexpected and unwanted effects. To avoid this, they are to be tested and optimized in extensive listening experiments.

It can be seen from fig. 1 that signals with a frequency lower than the frequency of the masking signal are hardly masked. Therefore only two kinds of masking are considered: in-band masking, which is masking within a subband, and masking of signals in subbands at higher frequencies. For both cases the masked power is computed as a function of the powers of the subband signals.

Firstly it is assumed that there is only one signal in the subband with index i. This subband ranges from frequencies $f_{l,i}$ to $f_{u,i}$. The signal power is $\sigma_{s,i}^2$. The quantization noise is assumed to have a flat spectrum in the subband. The worst-case situation for in-band masking occurs when the masking signal is a pure tone with a frequency $f_{u,i}$. In this case the power of the quantization noise in subband i that is masked by a signal with power $\sigma_{s,i}^2$ in the same subband must be less than $\sigma_{s,i}^2 T(f_{u,i}, f_{l,i})$. This situation is illustrated in fig. 6

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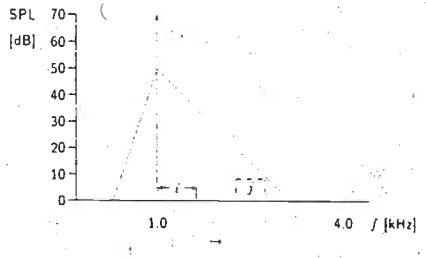


Fig. 7. Masking of subbands at higher frequencies.

for a pure tone of 1000 Hz at an SPL of 70 dB. In this figure subband i ranges from 830 to 1000 Hz.

The worst-case situation for the masking of noise in subbands at higher frequencies occurs when the masking signal in subband i is a pure tone with a frequency $f_{l,i}$. In this case the power of the quantization noise in subband j that is masked by a signal with power $\sigma_{x,i}^2$ in subband i must be less than $\sigma_{x,i}^2 T(f_{l,i}, f_{u,j})$. This situation is illustrated in fig. 7 for a pure tone of 1000 Hz at an SPL of 70 dB. In this figure subband i ranges from 1000 to 1330 Hz, and subband j ranges from 1780 to 2180 Hz.

In this way the contribution of a subband to the masked power in all subbands can be computed. Because masking is assumed to be additive, the masked power in a subband can be obtained by adding all contributions.

The lowest achievable bit rate at a certain quality level depends on the division into subbands. The computations of the masked powers are based on worst-case assumptions. The real masked powers can be substantially higher. If the subbands are narrower the results of these computations will, on average, be closer to the real masked powers. This effect, however, is limited for in-band masking, because in reality the tops of the curves of figs 5, 6, and 7 are flatter than is depicted [4]), so that there is no point in decreasing the bandwidth of the subbands below a certain value. Because of the effect mentioned here, narrower subbands lead to higher masked powers and consequently the number of bits required for quantization can be lower.

The results of this section are only valid if the distribution of signal power over the subbands is stationary. In reality this is not true. Therefore, in a subband coding system, the masked powers must be computed periodically. As a consequence of this nonstationarity the number of bits needed to quantize each subband in such a way that the quantization noise is masked will also

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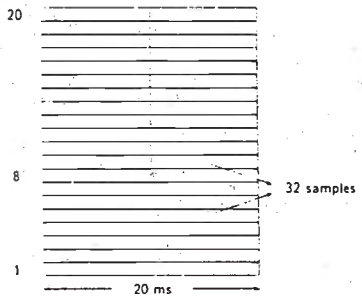


Fig. 8. Allocation window for a 20-band system.

vary in time. The allocation of bits to the quantizers in such a way that the coding system produces a fixed bit rate is discussed in Sec. 5.

5. Adaptive bit allocation

Before the subband signals are quantized they are divided into blocks. The blocks are arranged in an allocation window. An allocation window contains all subband samples during a period of time. This period is chosen in such a way that it contains one block of samples from the most decimated subband. This is in general the subband at the lowest frequency. An example of an allocation window for a 20-band system is shown in fig. 8. If the input sample frequency is 44 100 Hz, subbands 1–8 have a bandwidth of 689 Hz, and subbands 9–20 have a bandwidth of 1378 Hz.

An allocation window contains MD_{max} samples, where M is the quantization block length, and D_{max} is the maximum decimation factor. In the example of fig. 8, the block length and the maximum decimation factor both equal 32. The allocation window corresponds to a time duration T_w given by

$$T_w = \frac{MD_{max}}{f_s},$$

where f_s is the input sampling frequency. If the desired bit rate at the output is R, then $T_w R$ bits must be divided over the allocation window. A certain number of bits is reserved for peak values and side information. Good results have been obtained with peak values logarithmically quantized with 6 bits

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and 4 bits of side information per block indicating the number of bits per sample used for the block. Assume that B_q bits are left for quantization of the subband samples. The following adaptive bit allocation procedure is used to divide these bits over the allocation window.

First for each block in the allocation window the peak value and the power are computed. The power in a block is obtained as the average of the squares of the samples. By using the masking model of Sec. 4, the masked powers are now computed for every block in the allocation window. This is done in such a way that a block only contributes to the masked power in blocks that lie within the time interval of the masking block. For instance, the block in subband 1 of fig. 8 contributes to the masked powers of all blocks in the allocation window, the block in subband 3 contributes to the masked power of all blocks in subbands 3–20, and the left-hand block of subband 9 only contributes to the masked powers in the left-hand blocks of subbands 9–20.

Assume that the number of blocks in the allocation window is N and that they are numbered from 1 to N. The estimated masked power in the ith block is denoted by $\sigma_{m,i}^2$, and the peak value by p_i. If the samples in the block are uniformly quantized to b_i bits, then the power of the quantization noise in the ith block is assumed to be given by [1])

$$\frac{1}{12}\left(\frac{2p_i}{2^{b_i}}\right)^2.$$

The noise-to-mask ratio in the same block is defined by

$$\frac{1}{12}\left(\frac{2p_i}{2^{b_i}}\right)^2 \frac{1}{\sigma_{m,i}^2}.$$

In the following it is sometimes more convenient to use a logarithmic noise-to-mask ratio γ_i given by

$$\gamma_i = \log_2\left(\frac{p_i}{\sigma_{m,i}}\right) - b_i.$$

The adaptive bit allocation procedure is such that the total noise-to-mask ratio, given by

$$\sum_{i=1}^{N} \frac{1}{12}\left(\frac{2p_i}{2^{b_i}}\right)^2 \frac{1}{\sigma_{m,i}^2}$$

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I is minimized under the constraint

$$\sum_{i=1}^{N} Mb_i = B_q.$$

A further constraint is that all b_i must be integers with $0 \leqslant b_i \leqslant 15$.

An elegant solution to this constrained integer minimization problem can be derived from the theory given in ref. 12. The result of this procedure for this case, as described below, is intuitively pleasing. Initially, it is assumed that all blocks are quantized with zero bits. The number of bits left to divide over the allocation window then is given by

$$B = B_q,$$

and the logarithmic noise-to-mask ratio for the ith block by

$$\gamma_i = \log_2\left(\frac{p_i}{\sigma_{m,i}}\right).$$

Then the following three steps are repeated until $B < M$, so that no more bits can be assigned to the blocks:

1. the block i with the highest logarithmic noise-to-mask ratio γ_i is searched,
2. then the number of bits per sample b_i for this block is increased by one, and
3. the number of bits left B is decreased by M.

If this procedure is started with a sufficient number of bits, the result is that all noise-to-mask ratios will be approximately equal. The procedure can be refined by setting lower bounds to the noise-to-mask ratios of the subband. The reason for doing this is that once the noise-to-mask ratio in a block is below a certain threshold it is no longer necessary to add bits to the block because the quantization errors have become inaudible. In this way bits can be saved for other blocks.

Note that, if B_q is not a multiple of M, a small number of bits are left over, and the bit rate will be a fraction less than the desired bit rate R.

6. Results

The methods explained in this paper have been applied in two coding systems: a complex system splitting up the signal into 26 subbands,

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approximately one-third of an octave wide, and a simpler 20-band system of which the bandwidths were given in Sec. 5. In both systems the adaptive bit allocation method described in Sec. 5 has been used. Both systems have been designed for coding stereophonic 16-bit compact disc signals with a sample frequency of 44.1 kHz. Left and right channels are coded independently. With the 26-band system high-quality results can be obtained at bit rates of 200 kbit s^{-1}, which comes down to a reduction by a factor of seven. With the 20-band system similar results can be obtained at bit rates of 300 kbit s^{-1}, which comes down to a reduction by a factor of almost five. In listening experiments most of the processed pieces of music cannot be distinguished from the original source material. Physically, however, signal-to-noise ratios are measured between 15 and 30 dB, showing that the signal is heavily distorted.

The complexity of the systems is largely determined by the memory requirements of the filter banks. These are substantially higher for the 26-band system.

The filtering and coding delay is determined by the maximum decimation factor, the filter banks used, and the quantization block length. For the 26-band system the maximum decimation factor is 256 and the total delay can be as high as 800 ms. For the 20-band system the maximum decimation factor is 32 and a typical value for the total delay is 80 ms. Other types of filter banks and shorter quantization blocks may give lower values.

7. Conclusions

A subband coding method for high-quality digital audio signals has been described. It exploits a perceptive phenomenon called simultaneous masking. This phenomenon has been described with some examples. A masking model has been derived on which the coding algorithm is based. With this masking model the encoder computes the amount of quantization noise that is masked in each subband. An adaptive bit allocation algorithm distributes the bits over the subbands in such a way that the bit rate is fixed, and that the total noise-to-mask ratio is minimized. After decoding there is no, or very little, perceptible loss in quality, even at low bit rates.

Further improvements are possible. The masking model that is used can be extended in several ways. For instance, it can be adapted in such a way that it better describes masking at frequencies below 500 Hz. Temporal masking, which is masking of a signal occuring in time before or after the masking signal [2]), can also be incorporated. A further reduction in bit rate can be achieved by exploiting redundancies in the quantized subband samples.

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for instance by using a Huffman code[1]). Currently it is being investigated whether the bit rate can be further reduced by combined coding of the left and right channels of a stereophonic signal.

Acknowledgement

The authors would like to thank their colleague Gertjan Keesman for his help in developing the adaptive bit-allocation technique.

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Authors



Raymond Veldhuis: Ir. degree (Electrical Engineering), University of Twente, Enschede, The Netherlands, 1981; Ph.D., University of Nijmegen, The Netherlands, 1988; Philips Research Laboratories, Eindhoven, 1982– . He has been working on restoration methods for lost samples in audio, speech, and television signals. Currently he is investigating source coding methods for high quality digital audio systems.



Marcel Breeuwer: Ir. degree (Electrical Engineering), Technical University, Delft, The Netherlands, 1982; Ph.D., Free University of Amsterdam, 1985; Philips Research Laboratories, Eindhoven, 1985– . His Ph.D. work was on the effect of supplementing speech reading with auditory information. More recently he has been involved in source coding for high-quality digital audio signals. Currently he is working on video source coding.

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Robbert van der Waal: Ir. degree (Electrical Engineering), University of Twente, Enschede, The Netherlands, 1987; Philips Research Laboratories, Eindhoven, 1987– . He has been working on source coding methods for high-quality digital audio signals. Currently he is involved in the design of integrated analogue filters.

0129

Transaction History Date 1995-08-17
Date information retrieved from USPTO Patent Application Information Retrieval (PAIR) system records at www.uspto.gov

Atty Dkt.
03-48683

Customer Correction

PATENT

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

RECEIVED
MAY 21 1996
FEB 01 1996
GROUP 2100

Applicants: TEH, et al.

Serial No.: 08/161,798

Filed : December 6, 1993 Group Art Unit: 2104

For : A BIT ALLOCATION METHOD FOR TRANSFORM CODER

REQUEST FOR CORRECTED FILING RECEIPT

RECEIVED
FEB 06 1996

Honorable Commissioner
of Patents and Trademarks
Washington, D.C. 20231

Attention:
Application Processing Division
Customer Correction Branch

Dear Sir:

Applicants hereby request correction of the official Filing Receipt (copy enclosed) in the above-identified application, since the Filing Receipt incorrectly shows the Applicants as "Do *H*. Teh," "Pek *Y*. Tan" and Sua *H*. Neo." Please correct the Applicants' names to:

Do Hui TEH

Pek Yew TAN

Sua Hong NEO

as shown in the enclosed copy of the Declaration and Power of Attorney form, filed with the application on December 6, 1993.

The undersigned respectfully awaits receipt of a corrected Filing Receipt.

Respectfully submitted,

BEVERIDGE, DeGRANDI, WEILACHER & YOUNG

By: Joseph A. DeGrandi

Joseph A. DeGrandi, Reg. No. 17,446
1850 M Street, N.W., Suite 800
Washington, D.C. 20036
Telephone: (202) 659-2811

August 17, 1995

PTO-103X
(Rev. 7-93)

FILING RECEIPT

UNITED STATES DEPARTMENT OF COMMERCE
Patent and Trademark Office
ASSISTANT SECRETARY AND COMMISSIONER
OF PATENTS AND TRADEMARKS
Washington, D.C. 20231

APPLICATION NUMBER	FILING DATE	GRP ART UNIT	FIL FEE REC'D	ATTORNEY DOCKET NO.	DRWGS	TOT CL	IND CL
08/161,798	12/06/93	2104	$710.00	0348683	8	4	1

BEVERIDGE, DEGRANDI, WEILACHER & YOUNG
1850 M STREET, N. W., SUITE 800
WASHINGTON, DC 20036

Receipt is acknowledged of this patent application. It will be considered in its order and you will be notified as to the results of the examination. Be sure to provide the U.S. APPLICATION NUMBER, FILING DATE, NAME OF APPLICANT, and TITLE OF INVENTION when inquiring about this application. Fees transmitted by check or draft are subject to collection. Please verify the accuracy of the data presented on this receipt. If an error is noted on this Filing Receipt, please write to the Application Processing Division's Customer Correction Branch within 10 days of receipt. Please provide a copy of the Filing Receipt with the changes noted thereon.

Applicant(s) DO H. TEH, SINGAPORE, SINGAPORE; PEK Y. TAN, SINGAPORE, SINGAPORE; SUA H. NEO, SINGAPORE, SINGAPORE.

FOREIGN/PCT APPLICATIONS-JAPAN 5-030712 02/19/93

TITLE
BIT ALLOCATION METHOD FOR TRANSFORM CODER

PRELIMINARY CLASS: 341

(see reverse)

...ese Language Declaration

私は、合衆国法典第35部第119 条にもとづく下記の外国特許出願または発明者証出願の外国優先権利益を主張し、さらに優先権の主張に係わる基礎出願の出願日前の出願日を有する外国特許出願または発明者証出願を以下に明記する：

I hereby claim foreign priority benefits under Title 35, United States Code, §119 of any foreign application(s) for patent or inventor's certificate listed below and have also identified below any foreign application for patent or inventor's certificate having a filing date before that of the application on which priority is claimed:

Prior foreign applications
先の外国出願

(Number) (番 号)	(Country) (国 名)	(Day/Month/Year Filed) (出願の年月日)	Priority claimed 優先権の主張 Yes あり	No なし
P.appln.hei 5-30,712	Japan	19, February, 1993	☒	☐
			☐	☐
			☐	☐

私は、合衆国法典第35部第120 条にもとづく下記の合衆国特許出願の利益を主張し、本願の請求の範囲各項に記載の主題が合衆国法典第35部第112 条第１項に規定の態様で先の合衆国出願に開示されていない限度において、先の出願の出願日と本願の国内出願日またはＰＣＴ国際出願日の間に公表された連邦規則法典第37部第１章第56条（ａ）項に記載の所要の情報を開示すべき義務を有することを認める：

I hereby claim the benefit under Title 35, United States Code, §120 of any United States application(s) listed below and, insofar as the subject matter of each of the claims of this application is not disclosed in the prior United States application in the manner provided by the first paragraph of Title 35, United States Code, §112, I acknowledge the duty to disclose material information as defined in Title 37, Code of Federal Regulations, §1.56(a) which occurred between the filing date of the prior application and the national or PCT international filing date of this application:

(Application Serial No.) (出願番号)	(Filing Date) (出願日)	(現 況) (特許済み、係属中、放棄済み)	(Status) (patented, pending, abandoned)

私は、ここに自己の知識にもとづいて行った陳述がすべて真実であり、自己の有する情報および信ずるところに従って行った陳述が真実であると信じ、さらに故意に虚偽の陳述等を行った場合、合衆国法典第18部第1001条により、罰金もしくは禁錮に処せられるか、またはこれらの刑が併科され、またかかる故意による虚偽の陳述が本願ないし本願に対して付与される特許の有効性を損うことがあることを認識して、以上の陳述を行ったことを宣言する。

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 and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

...ese Language Declaration

委任状：私は、下記発明者として、以下の代理人をここに選任し、本願の手続を遂行すること並びにこれに関する一切の行為を特許商標庁に対して行うことを委任する。
（代理人氏名および登録番号を明記のこと）

POWER OF ATTORNEY: As a named inventor, I hereby appoint the following attorney(s) and/or agent(s) to prosecute this application and transact all business in the Patent and Trademark Office connected therewith. (*list name and registration number*)

Joseph A. DeGrandi (17446)
Robert G. Weilacher (20531)
Richard G. Young (20628)
John T. Roberts (20908)
Michael A. Makuch (32263)
Helen M. McCarthy (32513)
Dennis C. Rodgers (32936)
William F. Rauchholz (34701)
G. Byron Stover (34737)

書類の送付先：

Send Correspondence to:

BEVERIDGE, DEGRANDI, WEILACHER & YOUNG
1850 M Street, N.W., Suite 800
Washington, D.C. 20036

直通電話連絡先：（名称および電話番号）

Direct Telephone Calls to: (*name and telephone number*)

BEVERIDGE, DEGRANDI, WEILACHER & YOUNG 659-2811

唯一のまたは第一の発明者の氏名	Full name of sole or first inventor: Teh, Do Hui
同発明者の署名 日付	Inventor's signature: [signature] Date: 4 Oct 93
住所	Residence: Singapore
国籍	Citizenship: Malaysia
郵便の宛先	Post Office Address: Blk 119 Tampines Court Tampines St.11
	No.12-180 Singapore 1852
第2の共同発明者の氏名（該当する場合）	Full name of second joint inventor, if any: Tan, Pek Yew
同第2発明者の署名 日付	Second Inventor's signature: [signature] Date: 4 Oct 93
住所	Residence: Singapore
国籍	Citizenship: Malaysia
郵便の宛先	Post Office Address: Blk 128, Yishun St.11
	No.05-305 Singapore 2776

（第六またはそれ以降の共同発明者に対しても同様な情報および署名を提供すること。）

(Supply similar information and signature for third and subsequent joint inventors.)

Form PTO-FB-265 (8-83) Patent and Trademark Office-U.S. DEPARTMENT OF COMMERCE

第3の共同発明者の氏名（該当する場合）	Full name of third joint inventor, if any Neo, Sua Hong
同第3発明者の署名　日付	Third Inventor's signature　Date 4 Oct 95
住所	Residence Singapore
国籍	Citizenship Singapore
郵便の宛先	Post Office Address Blk 456, Ang Mo Kio Avenue 10
	No.03-1558 Singapore 2056
第4の共同発明者の氏名（該当する場合）	Full name of fourth joint inventor, if any
同第4発明者の署名　日付	Fourth Inventor's signature　Date
住所	Residence
国籍	Citizenship
郵便の宛先	Post Office Address
第5の共同発明者の氏名（該当する場合）	Full name of fifth joint inventor, if any
同第5発明者の署名　日付	Fifth Inventor's signature　Date
住所	Residence
国籍	Citizenship
郵便の宛先	Post Office Address
第6の共同発明者の氏名（該当する場合）	Full name of sixth joint inventor, if any
同第6発明者の署名　日付	Sixth Inventor's signature　Date
住所	Residence
国籍	Citizenship
郵便の宛先	Post Office Address
第7の共同発明者の氏名（該当する場合）	Full name of seventh joint inventor, if any
同第7発明者の署名　日付	Seventh Inventor's signature　Date
住所	Residence
国籍	Citizenship
郵便の宛先	Post Office Address

2,08 2300 #4

DATE: 12-6-95

FROM: CUSTOMER CORRECTIONS
APPLICATION PROCESSING DIVISION
Loc. 0380

SUBJ: APPLICATION FILES NEEDED FOR CORRECTION/UPDATE

GROUP ART UNIT: ~~2307~~ 2300

APPLICATION NUMBER 08/161,788 IS NEEDED IMMEDIATELY FOR CORRECTION.

PLEASE ATTACH THIS FORM TO THE ABOVE APPLICATION AND RETURN IT TO THE APPLICATION PROCESSION DIVISION, CUSTOMER'S CORRECTION CP2- ROOM 6C17.

IF YOU ARE UNABLE TO LOCATE THE APPLICATION OR HAVE A QUESTION, PLEASE CALL ME AT 308-3618.

THANK YOU FOR YOUR ASSISTANCE

Norma White

NORMA WHITE
SUPERVISOR

Corrected

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MAY 21 1996
GROUP 2300

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FEB 01 1996
GROUP 2100

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FEB 06 1996
GROUP 2300

Transaction History Date 1996-06-17
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Patent and Trademark Office
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Washington, D.C. 20231

SERIAL NUMBER	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.
08/161,778	12/06/93	TEH	D 02485/69

EXAMINER

ART UNIT	PAPER NUMBER
2308	5

DATE MAILED: 06/17/96

BEVERIDGE, DEGRANDI, WEILACHER & YOUNG
1850 M STREET, N. W., SUITE 800
WASHINGTON, DC 20036

This is a communication from the examiner in charge of your application.
COMMISSIONER OF PATENTS AND TRADEMARKS

☑ This application has been examined ☐ Responsive to communication filed on _____ ☐ This action is made final.

A shortened statutory period for response to this action is set to expire 3 month(s), 0 days from the date of this letter. Failure to respond within the period for response will cause the application to become abandoned. 35 U.S.C. 133

Part I THE FOLLOWING ATTACHMENT(S) ARE PART OF THIS ACTION:

1. ☑ Notice of References Cited by Examiner, PTO-892.
2. ☑ Notice of Draftsman's Patent Drawing Review, PTO-948.
3. ☑ Notice of Art Cited by Applicant, PTO-1449.
4. ☐ Notice of Informal Patent Application, PTO-152.
5. ☐ Information on How to Effect Drawing Changes, PTO-1474.
6. ☐ _____

Part II SUMMARY OF ACTION

1. ☑ Claims 1-4 are pending in the application.
   Of the above, claims _____ are withdrawn from consideration.
2. ☐ Claims _____ have been cancelled.
3. ☐ Claims _____ are allowed.
4. ☑ Claims 1-4 are rejected.
5. ☐ Claims _____ are objected to.
6. ☐ Claims _____ are subject to restriction or election requirement.
7. ☐ This application has been filed with informal drawings under 37 C.F.R. 1.85 which are acceptable for examination purposes.
8. ☐ Formal drawings are required in response to this Office action.
9. ☐ The corrected or substitute drawings have been received on _____. Under 37 C.F.R. 1.84 these drawings are ☐ acceptable; ☐ not acceptable (see explanation or Notice of Draftsman's Patent Drawing Review, PTO-948).
10. ☐ The proposed additional or substitute sheet(s) of drawings, filed on _____. has (have) been ☐ approved by the examiner; ☐ disapproved by the examiner (see explanation).
11. ☐ The proposed drawing correction, filed _____, has been ☐ approved; ☐ disapproved (see explanation).
12. ☑ Acknowledgement is made of the claim for priority under 35 U.S.C. 119. The certified copy has ☑ been received ☐ not been received ☐ been filed in parent application, serial no. _____ ; filed on _____.
13. ☐ Since this application apppears to be in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under Ex parte Quayle, 1935 C.D. 11; 453 O.G. 213.
14. ☐ Other

Part III DETAILED ACTION

Drawings

1. This application has been filed with informal drawings which are acceptable for examination purposes only. Formal drawings will be required when the application is allowed.

Specification

2. The title of the invention is not descriptive. A new title is required that is clearly indicative of the invention to which the claims are directed.

Claim Objections

3. Claims 1-2 objected to because of the following informalities.

As per Claim 1, capitalize first letter "a".

As per Claim 2, line 2 "quantizations is that" is incorrect grammar.

Appropriate correction is required.

Claim Rejections - 35 USC § 112

4. Claims 1-4 rejected under 35 U.S.C. § 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which applicant regards as the invention.

As per Claim 1,

a) lines 2-3, "the audio signals" lacks clear antecedent basis, should this be "the digital audio signals;"

b) line 5, "a variance and/or a representative" is vague and indefinite and is alternative language which are not connected;

c) line 7-8, "a psychoacoustics criteria" is indefinite in that "psychoacoustics" should be singular;

d) line 9, "an initial quantizations" is indefinite in that "quantizations" should be singular;

e) line 11, "iteratively increasing quantizations" is vague and indefinite in that this phrase does not explain what is being quantized; further, "increasing quantizations" has no concrete meaning and is hanging;

f) lines 11-12, "necessary and final bit rate" is vague and indefinite and lacks antecedent basis; please explain these terms further in the claim;

g) line 12, "maintaining effects" is vague and indefinite and does not logically connect to "said model";

h) line 12, "said model" lacks clear antecedent basis.

As per Claim 2,

a) line 3, "iteratively increasing quantizations" is vague and indefinite; what is being quantized and by what?

As per Claim 3,

a) lines 2-3, "said variance or representative" is vague and indefinite and alternative language;

b) line 5, "taken to be the bandwidth" is vague and indefinite and does not connect to the rest of the claim.

As per Claim 4,

a) line 2, "in consideration" is vague and does not clearly recite how a mathematical model is derived from in-band masking effect.

Claim Rejections - 35 USC § 102

5. The following is a quotation of the appropriate paragraphs of 35 U.S.C. § 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless --
> (b) the invention was patented or described in a printed publication in this or a foreign country or in public use or on sale in this country, more than one year prior to the date of application for patent in the United States.

6. Claims 1-4 rejected under 35 U.S.C. § 102(b) as being anticipated by Veldhuis et al. (5,105,463).

As per Claim 1, Veldhuis teaches

a) audio signals buffered in frames and decomposed into its spectral components (his Fig 1, block 6(1) to 6(26) and "allocation windows" which are similar to frames, col. 2 lines 40-50);

b) obtaining a variance or a representative of the components of said frame within a defined frequency interval (his comparator means col. 2, lines 53-63);

c) computing a necessary bandwidth using a psychoacoustic criteria (his col 1, lines 20-26);

d) determining an initial quantizations of each said frequency interval using an approximate mathematical model (his col 3, lines 3-29 with mathematical algorithm being given by B(p) and G(p;m) and C(p;m));

e) iteratively increasing quantizations for satisfying necessary and final bit rate, maintaining effects of said model (his col 3, lines 19-29);

As per Claim 2, Veldhuis teaches iteratively increasing quantizations of each said frequency interval through a psychoacoustically weighted inverse relationship with the said variance of each said frequency interval (his col 2, lines 57-68

and col 3, lines 1-29 in which inverse relationship for increasing bit allocation is given in col 3, lines 10-29).

As per Claim 3, Veldhuis teaches the bandwidth being computed by comparing the said variance or representative against a human hearing threshold such that the largest frequency interval with said variance or representative exceeding the hearing threshold taken to be the bandwidth (his Abstract, first sentence further explained on col 1, lines 20-26).

As per Claim 4, Veldhuis teaches said mathematical model is derived in consideration of an in-band masking effect such that an error spectrum will follow more closely to the original signal spectrum (his col 2, lines 54-64 in which mathematical algorithm uses thresholds of human auditory system for just perceiving single tones).

Conclusion

7. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

Nishiguchi et al. (5,151,941) teaches transform coder in which coding is done based on allowable noise level.

Veldhuis ("Subband Coding of Digital Audio Signals Without Loss of Quality") teaches exploiting the simultaneous masking effect of the human ear in an adaptive bit allocation scheme.

8. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Indranil Chowdhury whose telephone number is (703) 305-3868.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Allen R. MacDonald, can be reached on (703) 305-9708. The facsimile phone number for this group is (703) 308-5356.

Any inquiry of a general nature or relating to the status of this application should be directed to the Group receptionist whose telephone number is (703) 305-9600.

ALLEN R. MACDONALD
SUPERVISORY PATENT EXAMINER
ART UNIT 2308

IC
June 10, 1996

TO SEPARATE, HOLD TOP AND BOTTOM EDGES, SNAP-APART AND DISCARD CARBON

FORM PTO-892 (REV. 2-92)	U.S. DEPARTMENT OF COMMERCE PATENT AND TRADEMARK OFFICE	SERIAL NO.	GROUP ART UNIT	ATTACHMENT TO PAPER NUMBER
NOTICE OF REFERENCES CITED		08/161,798	2308	5
		APPLICANT(S) Teh et al.		

U.S. PATENT DOCUMENTS

*		DOCUMENT NO.	DATE	NAME	CLASS	SUB-CLASS	FILING DATE IF APPROPRIATE
	A	5105463	4/1992	Veldhuis et al.	381	30	
	B	5151941	9/1992	Nishiguchi et al.	381	46	
	C						
	D						
	E						
	F						
	G						
	H						
	I						
	J						
	K						

FOREIGN PATENT DOCUMENTS

*		DOCUMENT NO.	DATE	COUNTRY	NAME	CLASS	SUB-CLASS	PERTINENT SHTS. DWG	PERTINENT PP. SPEC.
	L								
	M								
	N								
	O								
	P								
	Q								

OTHER REFERENCES (Including Author, Title, Date, Pertinent Pages, Etc.)

R	Veldhuis et al, "Subband Coding of Digital Audio Signals Without Loss of Quality," ICASSP-89, 1989, pp. 2009-2012
R	

EXAMINER	DATE
Indranil Chowdhury	6/10/96

* A copy of this reference is not being furnished with this office action.
(See Manual of Patent Examining Procedure, section 707.05 (a).)

Other Prior Art

According to the information contained in form PTO-1449 or PTO-892, there are one or more other prior art/non-patent literature documents missing from the original file history record obtained from the United States Patent and Trademark Office. Upon your request we will attempt to obtain these documents from alternative resources. Please note that additional charges will apply for this service.

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Form PTO 948 (Rev. 10-93) U.S. DEPARTMENT OF COMMERCE - Patent and Trademark Office Application No. 16/798

NOTICE OF DRAFTSPERSON'S PATENT DRAWING REVIEW

PTO Draftspersons review all originally filed drawings regardless of whether they are designated as formal or informal. Additionally, patent Examiners will review the drawings for compliance with the regulations. Direct telephone inquiries concerning this review to the Drawing Review Branch, 703-305-8404.

The drawings filed (insert date) 12/6/93, are
A ___ not objected to by the Draftsperson under 37 CFR 1.84 or 1.152.
B ___ objected to by the Draftsperson under 37 CFR 1.84 or 1.152 as indicated below. The Examiner will require submission of new, corrected drawings when necessary. Corrected drawings must be submitted according to the instructions on the back of this Notice.

1. DRAWINGS. 37 CFR 1.84(a): Acceptable categories of drawings: Black ink. Color.
   ___ Not black solid lines. Fig(s)___
   ___ Color drawings are not acceptable until petition is granted.

2. PHOTOGRAPHS. 37 CFR 1.84(b)
   ___ Photographs are not acceptable until petition is granted.

3. GRAPHIC FORMS. 37 CFR 1.84 (d)
   ___ Chemical or mathematical formula not labeled as separate figure. Fig(s)___
   ___ Group of waveforms not presented as a single figure, using common vertical axis with time extending along horizontal axis. Fig(s)___
   ___ Individuals waveform not identified with a separate letter designation adjacent to the vertical axis. Fig(s)___

4. TYPE OF PAPER. 37 CFR 1.84(e)
   ___ Paper not flexible, strong, white, smooth, nonshiny, and durable. Sheet(s)___
   ___ Erasures, alterations, overwritings, interlineations, cracks, creases, and folds not allowed. Sheet(s)___

5. SIZE OF PAPER. 37 CFR 1.84(f): Acceptable paper sizes:
   21.6 cm. by 35.6 cm. (8 1/2 by 14 inches)
   21.6 cm. by 33.1 cm. (8 1/2 by 13 inches)
   21.6 cm. by 27.9 cm. (8 1/2 by 11 inches)
   21.0 cm. by 29.7 cm. (DIN size A4)
   ___ All drawing sheets not the same size. Sheet(s)___
   ___ Drawing sheet not an acceptable size. Sheet(s) ___

6. MARGINS. 37 CFR 1.84(g): Acceptable margins:

Paper size

	21.6 cm. X 35.6 cm. (8 1/2 X 14 inches)	21.6 cm. X 33.1 cm. (8 1/2 X 13 inches)	21 cm. X 27.9 cm. (8 1/2 X 11 inches)	21 cm. X 29.7 cm. (DIN Size A4)
T	5.1 cm. (2")	2.5 cm. (1")	2.5 cm. (1")	2.5cm.
L	.64 cm. (1/4")	.64 cm. (1/4")	.64 cm. (1/4")	2.5 cm.
R	.64 cm. (1/4")	.64 cm. (1/4")	.64 cm. (1/4")	1.5 cm.
B	.64 cm. (1/4")	.64 cm. (1/4")	.64 cm. (1/4")	1.0 cm.

   Margins do not conform to chart above.
   Sheet(s)___
   ___Top (T) ___ Left (L) ___Right (R) ___Bottom (B)

7. VIEWS. 37 CFR 1.84(h)
   REMINDER: Specification may require revision to correspond to drawing changes.
   ___ All views not grouped together. Fig(s)___
   ___ Views connected by projection lines. Fig(s)___
   ___ Views contain center lines. Fig(s)___

Partial views. 37 CFR 1.84(h)(2)
   ___ Separate sheets not linked edge to edge. Fig(s)___
   ___ View and enlarged view not labeled separately. Fig(s)___
   ___ Long view relationship between different parts not clear and unambiguous. 37 CFR 1.84(h)(2)(ii) Fig(s)___

Sectional views. 37 CFR 1.84(h)(3)
   ___ Hatching not indicated for sectional portions of an object. Fig(s)___
   ___ Hatching of regularly spaced oblique parallel lines not spaced sufficiently. Fig(s)___
   ___ Hatching not at substantial angle to surrounding axes or principal lines. Fig(s)___
   ___ Cross section not drawn same as view with parts in cross section with regularly spaced parallel oblique strokes. Fig(s)___
   ___ Hatching of juxtaposed different elements not angled in a different way. Fig(s)___

Alternate position. 37 CFR 1.84(h)(4)
   ___ A separate view required for a moved position. Fig(s)___

Modified forms. 37 CFR 1.84(h)(5)
   ___ Modified forms of construction must be shown in separate views. Fig(s)___

8. ARRANGEMENT OF VIEWS. 37 CFR 1.84(i)
   ___ View placed upon another view or within outline of another. Fig(s)___
   ___ Words do not appear in a horizontal, left-to-right fashion when page is either upright or turned so that the top becomes the right side, except for graphs. Fig(s)___

9. SCALE. 37 CFR 1.84(k)
   ___ Scale not large enough to show mechanism without crowding when drawing is reduced in size to two-thirds in reproduction. Fig(s)___
   ___ Indication such as "actual size" or "scale 1/2" not permitted. Fig(s)___
   ___ Elements of same view not in proportion to each other. Fig(s)___

10. CHARACTER OF LINES, NUMBERS, & LETTERS. 37 CFR 1.84(l)
   ___ Lines, numbers & letters not uniformly thick and well defined, clean, durable, and black (except for color drawings). Fig(s) 1-8

11. SHADING. 37 CFR 1.84(m)
   ___ Shading used for other than shape of spherical; cylindrical, and conical elements of an object, or for flat parts. Fig(s)___
   ___ Solid black shading areas not permitted. Fig(s)___

12. NUMBERS, LETTERS, & REFERENCE CHARACTERS. 37 CFR 1.84(p)
   ___ Numbers and reference characters not plain and legible. 37 CFR 1.84(p)(l) Fig(s) 1-8
   ___ Numbers and reference characters used in conjuction with brackets, inverted commas, or enclosed within outlines. 37 CFR 1.84(p)(l) Fig(s)___
   ___ Numbers and reference characters not oriented in same direction as the view. 37 CFR 1.84(p)(l) Fig(s)___
   ___ English alphabet not used. 37 CFR 1.84(p)(2) Fig(s)___
   ___ Numbers, letters, and reference characters do not measure at least .32 cm. (1/8 inch) in height. 37 CFR(p)(3) Fig(s) 1-8

13. LEAD LINES. 37 CFR 1.84(q)
   ___ Lead lines cross each other. Fig(s)___
   ___ Lead lines missing. Fig(s)___
   ___ Lead lines not as short as possible. Fig(s)___

14. NUMBERING OF SHEETS OF DRAWINGS. 37 CFR 1.84(t)
   ___ Number appears in top margin. Fig(s)___
   ___ Number not larger than reference characters. Fig(s)___
   ___ Sheets not numbered consecutively, and in Arabic numerals, beginning with number 1. Sheet(s)___

15. NUMBER OF VIEWS. 37 CFR 1.84(u)
   ___ Views not numbered consecutively, and in Arabic numerals, beginning with number 1. Fig(s)___
   ___ View numbers not preceded by the abbreviation Fig. Fig(s)___
   ___ Single view contains a view number and the abbreviation Fig.
   ___ Numbers not larger than reference characters. Fig(s)___

16. CORRECTIONS. 37 CFR 1.84(w)
   ___ Corrections not durable and permanent. Fig(s)___

17. DESIGN DRAWING. 37 CFR 1.152
   ___ Surface shading shown not appropriate. Fig(s)___
   ___ Solid black shading not used for color contrast. Fig(s)___

0145

ATTACHMENT TO PAPER NO.___ REVIEWER___ DATE 3/8/94

Attorney Docket No. 03/58-48683

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicant : TEH, et al.

Serial No. 08/161,798 Examiner: I. Chowdhury

Filed December 6, 1993 Group Art Unit: 2308

For A BIT ALLOCATION METHOD FOR DIGITAL AUDIO SIGNALS (AS AMENDED)

MAIL ROOM OCT 17 1996 PAT. & TRADEMARK OFF.

Hon. Commissioner of Patents
and Trademarks
Washington, D.C. 20231

RECEIVED
OCT 30 1996
GROUP 2300

Sir:

TRANSMITTAL LETTER

Transmitted herewith for filing is:

Item	Fee
Amendment	$ 22.00
Petition for Extension of Time of 1 month	$110.00

For Amendment to Patent Application:

	Claims Remaining After		Highest No. Previously Paid For		Present Extra		SMALL ENTITY Rate	Addit Fee	OR	LARGE ENTITY Rate	Addit. Fee
Total	21	-	20	=	1	x	$11	$	x	$22	$22.00
Indep	1	-	3	=	--	x	$40	$	x	$80	$
							TOTAL	$	OR	TOTAL	$22.00

A check in the amount of $132.00 is attached hereto.

Please charge our Deposit Account No. 02-4300 for any other fees which may be required or credit any overpayment.

Respectfully submitted,

BEVERIDGE, DEGRANDI, WEILACHER & YOUNG, LLP

By: Joseph A DeGrandi

Joseph A. DeGrandi, Reg. No. 17,446
1850 M Street, N.W. (Suite 800)
Washington, D.C. 20036
Telephone: (202) 659-2811
Fax: (202) 659-1462

Date: October 17, 1996
Client No.: MAT 05

22-103
GP2308

Atty Docket
03/58-48683

PATENT

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

MAIL ROOM
OCT 17 1996
PAT & TRADEMARK OFF.

Applicant : TEH, et al.

Serial No. : 08/161,798 Examiner: I. Chowdhury

Filed : December 6, 1993 Group Art Unit: 2308

For : A BIT ALLOCATION METHOD FOR DIGITAL AUDIO SIGNALS (AS AMENDED)

AMENDMENT

RECEIVED
OCT 30 1996
GROUP 2300

Hon. Commissioner of Patents
and Trademarks
Washington, D.C. 20231

Sir:

This paper is submitted in response to the Office Action of June 17, 1996 (Paper No. 5). Also submitted herewith, together with the requisite fee, is a Petition for extending the time of response up to and including October 17, 1996. Please amend this application as follows.

IN THE TITLE:

Change "TRANSFORM CODER" to --DIGITAL AUDIO SIGNALS--.

IN THE ABSTRACT OF THE DISCLOSURE:

Line 1, change "of" to --for--, and after "having" insert --a--;

Line 3, delete "its";

Line 6, change "a psychoacoustics" to --psychoacoustic--;

Line 7, change "quantizations of each of the" to --quantization for each--;

1 103 22.00 DK

Line 8, change "quantizations of each" to --the quantization of each of--; and

Lines 9 and 10, change "interval" to --intervals--.

IN THE SPECIFICATION:

Page 3, line 4, change "quite a number" to --numerous examples--;

line 14, change "J D" to --J.D.--; and

line 20, change "N S" to --N.S.-- and change "P Noll" to -- P. Noll, which--.

Page 4, line 11, change "a full DSP's" to --full DSP--;

line 12, delete "processing"; and

line 21, change "his" to --this--.

Page 5, line 2, after "algorithm," insert --a--;

line 19, change "effect" to --affect--; and

line 21, after "has" insert --a--.

Page 6, line 2, after "realizes" insert --a--;

line 5, after "allocating" insert --an--;

line 7, change "interative" to --iterative--;

line 18, change "simplifies" to --simplifying--; and

line 22, after "is" insert --the--.

Page 7, line 2, change "such" to --the--;

lines 10, 12, and 14, change "flow chart" to --flowchart--;

line 19, change "(DB)" to --(dB)--; and

line 20, after "of" insert --a--.

Page 8, line 4, change "Firstly" to --First--;

line 6, delete "of" (first occurrence);

line 9, delete "of" (first occurrence);

line 10, delete "of";

line 16, delete "of" (first and second occurrences), and after "4," insert --the--;

line 18, delete "of" (second occurrence); and

line 19, delete "in".

Page 9, line 3, delete "of";

line 4, delete "of";

line 6, delete "of";

line 8, delete "of";

line 11, delete "of";

line 12, change "Step" to --Steps--, and change "is" to --are--; and

line 13, change "data(WL)" to --data (WL)--.

Page 10, line 1, after "by" insert --the--;

line 2, after "6" insert --,--, delete "of" (first occurrence), and change "of" (second occurrence) to --from--;

line 3, after "by" insert --an--;

line 4, change "6,7," to --6, 7,--;

line 7, change "procedure.A" to --procedure. A";

line 8, change "by" to --from--;

line 9, change "test" to --tests--;

line 11, change "Firstly" to --First--;

line 12, delete "of" (both occurrences), and delete "a";

line 15, delete "(";

line 16, change "step of numeral 33" to --(step numeral 33)--;

line 17, delete "of";

line 18, delete "of";

line 21, delete "of"; and

line 22, delete "of".

Page 11, line 1, delete "the";

line 5, change "Firstly" to --First--;

line 6, delete "of";

line 7, delete "of";

line 10, delete "of";

line 11, change "not all the units are" to --all of the units are not--;

line 13, delete "of"; and

line 17, change "psychoacoustics" to --psychoacoustic--.

Page 12, line 2, change "Steps of" to --Step--, and change "checks" to --check--;

line 3, change "is" to --are--, change "exceeds" to --exceed--, and delete "of";

line 4, delete "the" (first occurrence), change "limit(0)" to --limit (0 bits)--, and change "limit(15 bits) is" to

--limit (15 bits) are--;

line 14, change "show" to --shows--; and

line 21, after "to" (second occurrence) insert --the--.

Page 13, line 1, change "at least up to" to --up to at least--.

IN THE CLAIMS:

Please cancel claims 1-3.

Please amend claim 4 as follows:

Claim 4, line 1, change "1" to --5--; and

line 2, change "in consideration of" to --from--.

Please add new claims 5-24 as follows:

--5. 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:

a step for obtaining a representative of each frequency-unit obtained by dividing at least one defined frequency interval;

a step for counting a number of available frequency-units by using the representative, based on psychoacoustic

criteria, said available frequency-units satisfy said psychoacoustic criteria;

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

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

6. A method according to claim 5, wherein said second bit allocation step is to determine the bit allocation through a psychoacoustically weighted inverse relationship with the said representative of each of said frequency-units.

7. A method according to claim 6, wherein said counted number of available frequency-units is a number of the frequency-units which have the representative exceeding a human hearing threshold.

8. A method according to claim 7, wherein said mathematical model is represented by a formula:

$$WL(u) = \frac{var(u)}{F} - \frac{1}{N \cdot F} \sum_{u=0}^{N} var(u) + \frac{Ba}{A \cdot N}$$

wherein

WL(u) represents number of bits allocated to the frequency-units u,

var(u) represents variance of the frequency-units u,

F represents a psychoacoustic weighting factor,

N represents the counter number of the available frequency-units,

Ba represents number of bits set aside for initial allocation, and

A represents approximate average number of spectral samples in the available frequency-units.

9. Method according to claim 5, wherein each frequency-unit has a single spectral component.

10. Method according to claim 6, wherein each frequency-unit has a single spectral component.

11. Method according to claim 7, wherein each frequency-unit has a single spectral component.

12. Method according to claim 4, wherein each frequency-unit has a single spectral component.

13. Method according to claim 8, wherein each frequency-unit has a single spectral component.

14. Method according to claim 5, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.

15. Method according to claim 6, wherein said representative is a variance which is calculated from an

amplitude of a maximum spectral component of each frequency-unit.

16. Method according to claim 7, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.

17. Method according to claim 4, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.

18. Method according to claim 8, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.

19. Method according to claim 5, wherein each frequency-unit is obtained by dividing said spectral and temporal structure with at least one frequency interval.

20. Method according to claim 5, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.

21. Method according to claim 6, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.

22. Method according to claim 7, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.

23. Method according to claim 7, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.

24. Method according to claim 8, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.--

REMARKS

Reconsideration of this application is courteously solicited.

Claims 4-24 are pending in this application. New claims 5-7 are based on the subject matter originally claimed in cancelled claims 1-3 and discussed in the disclosure, for example, on pages 8-9 and 10-12. New claim 8 is based on the disclosure contained, for example, on page 11, equation 1. New claims 9-24 are based on the disclosure contained, for example, on page 8. Minor editorial amendments have been made to the specification and Abstract of the Disclosure hereby. Additional claim fees are required and submitted herewith.

With reference to the Office Action of June 17, 1996, the Examiner's acknowledgement of receipt of the priority papers has been noted.

It is proposed to amend Figure 3 to more clearly reflect "yes" and "no" branches from step numeral 42 as indicated on page 11, lines 7-13; and to correct a typographical error in step numeral 43. A hand corrected copy of Figure 3, with the

proposed addition and typographical error correction, shown in red, is submitted herewith. The Examiner's review and approval of the proposed amended Figure 3 are solicited.

The Office Action objected to the title as not descriptive of the claimed invention. As required, the title has been amended hereby to better describe the claimed invention, as a bit allocation method for digital audio signals.

Claims 1 and 2 have been objected to for informalities. The Examiner's cited informalities have been corrected in new claims 5 and 6, respectively.

Claims 1-4 have been rejected under 35 U.S.C. §112, second paragraph, for failing to particularly point out and distinctly claim the subject matter which Applicants regard as the invention. This rejection is moot with respect to cancelled claims 1-3. This rejection is traversed with respect to claim 4.

The amendment of claim 4 is believed to place claim 4 within the requirements of 35 U.S.C. §112. New claims 5-7, which replaced cancelled claims 1-3, are also believed to be in compliance with 35 U.S.C. §112.

Claims 1-4 have been rejected under 35 U.S.C. §102(b) for purportedly being anticipated by Veldhuis et al. This rejection is moot with respect to cancelled claims 1-3, but will be discussed with reference to new claims 5-7. This rejection is traversed with respect to claim 4.

Courts have stated that 35 U.S.C. §102(b) requires that a cited reference must contain all of the limitations of a claim

to be able to invalidate the claim based on anticipation. *See Elmer v. ICC Fabricating Inc.*, 67 F.3d 1571, 36 USPQ2d 1417, 1419 (Fed. Cir. 1995). In more general terms, 35 U.S.C. §102 taken as a whole requires that each and every element of a claim be present in a single cited reference to properly have that reference anticipate that claim. *See In re Bond*, 910 F.2d 831, 15 USPQ2d 1566, 1567 (Fed. Cir. 1992), *citing Diversitech Corp. v. Century Steps, Inc.*, 850 F.2d 675, 677, 7 USPQ2d 1315, 1317 (Fed. Cir. 1988), *Lindemann Maschinenfabrik v. American Hoist & Derrick Co.*, 730 F.2d 1452, 1458, 221 USPQ 481, 485 (Fed. Cir. 1984); *Minnesota Mining & Manufacturing Co. v. Johnson & Johnson Orthopaedics, Inc.*, 976 F.2d 1559, 24 USPQ2d 1321, 1326 (Fed. Cir. 1992).

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.

Arguendo, claim 4, as amended, provides for the mathematical model to be derived with respect to a known in-band masking effect. This reliance as shown in Applicants' Figure 5 for utilizing a threshold for the human auditory system to mask quantization noise at variety of frequency points as illustrated by the curve d is not the same basis for perceiving single tones as discussed in Veldhuis et al., col. 2, lines 54-64. Consequently, Veldhuis et al. do not disclose the reliance of a mathematical model on the in-band masking as claimed by Applicants.

In view of the above amendments and Remarks, it courteously is urged that all the claims are allowable and that the application now is in condition for allowance. Favorable action in this regard earnestly is solicited.

Respectfully submitted,

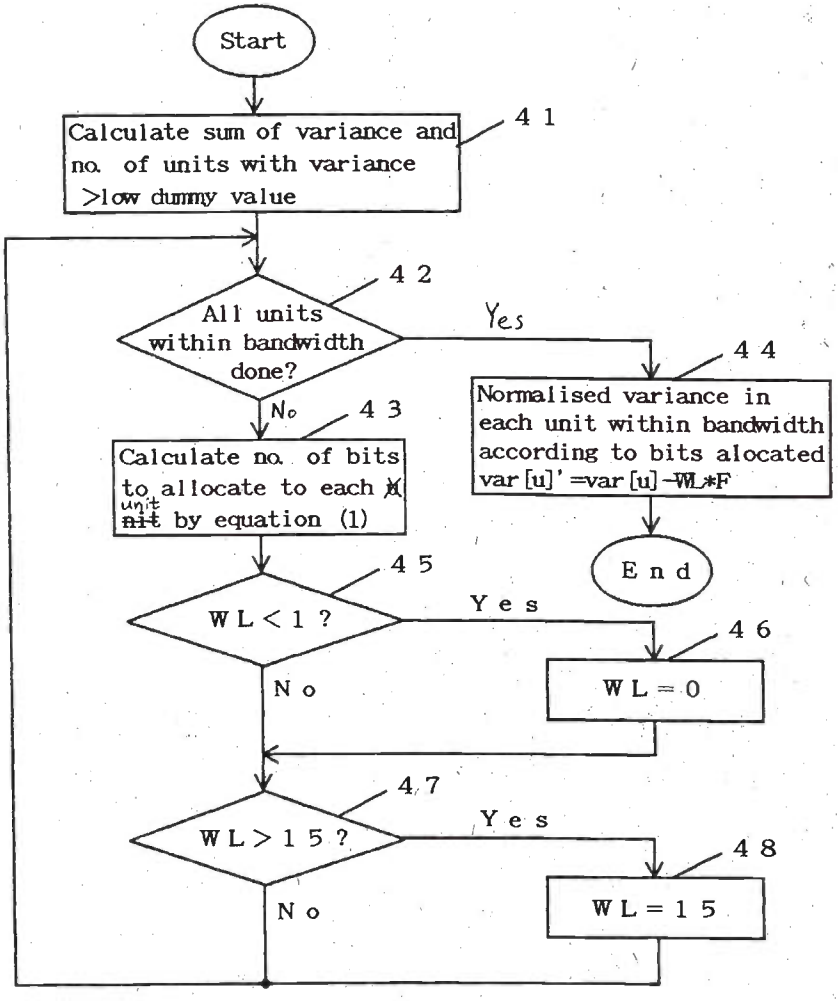
BEVERIDGE, DEGRANDI, WEILACHER & YOUNG, L.L.P.

By: Joseph A. DeGrandi

Joseph A. Degrandi, Reg. No. 17,446
1850 M Street, N.W. (Suite 800)
Washington, D.C. 20036
Telephone: (202) 659-2811
Fax: (202) 659-1462

October 17, 1996

Fig. 3


Start
Calculate sum of variance and no. of units with variance >low dummy value
41
All units within bandwidth done?
42
Yes
No
Calculate no. of bits to allocate to each unit by equation (1)
43
Normalised variance in each unit within bandwidth according to bits alocated var[u]' =var[u]-WL*F
44
End
WL < 1 ?
45
Yes
No
WL = 0
46
WL > 15 ?
47
Yes
No
WL = 15
48

110 - 115

08/13230

Atty Docket
03/58-48683

PATENT

#6
MDJ
11.1.96



IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicant : TEH, et al.

Serial No. : 08/161,798 Examiner: I. Chowdhury

Filed : December 6, 1993 Group Art Unit: 2308

For : A BIT ALLOCATION METHOD FOR TRANSFORM CC

PETITION FOR EXTENSION OF TIME

RECEIVED
OCT 30 1996
GROUP 2300

Hon. Commissioner of Patents
and Trademarks
Washington, D.C. 20231

Sir:

Applicants, through their undersigned attorney, res fully request a one month extension of the time period f responding to the outstanding Office Action dated June 1

With this filing, the period for response is thus e up to and including October 17, 1996.

Our check including the $110.00 extension of time fee is included herewith.

Please charge any additional fees which may be necessary to the Deposit Account of the undersigned, Deposit Account No. 02-4300, and credit any overpayment to said Deposit Account.

Respectfully submitted,

BEVERIDGE, DEGRANDI, WEILACHER & YOUNG, L.L.P.

By: Joseph A Degrandi

Joseph A. Degrandi, Reg. No. 17,446
1850 M Street, N.W. (Suite 800)
Washington, D.C. 20036
Telephone: (202) 659-2811
Fax: (202) 659-1462

October 17, 1996

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SERIAL NUMBER	FILING DATE	FIRST NAMED APPLICANT	ATTORNEY DOCKET NO.
08/161,798	12/06/93	TEH	D 0348683

E3M1/0106
BEVERIDGE, DEGRANDI, WEILACHER & YOUNG
1850 M STREET, N. W., SUITE 800
WASHINGTON DC 20036

EXAMINER	
GROVER, J	
ART UNIT	PAPER NUMBER
2308	8

DATE MAILED: 01/06/97

Please find below a communication from the EXAMINER in charge of this application.

Commissioner of Patents

see Attachments
12/26/96
VMG

PTOL-90 (Rev. 6/84)

1 - PATENT APPLICATION FILE COPY

Office Action Summary

Application No.	Applicant(s)
08/161,798	Teh et al.
Examiner	**Group Art Unit**
John Michael Grover	2308

☒ Responsive to communication(s) filed on *Oct 17, 1996*

☒ This action is **FINAL**.

☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11; 453 O.G. 213.

A shortened statutory period for response to this action is set to expire *3* month(s), or thirty days, whichever is longer, from the mailing date of this communication. Failure to respond within the period for response will cause the application to become abandoned. (35 U.S.C. § 133). Extensions of time may be obtained under the provisions of 37 CFR 1.136(a).

Disposition of Claims

☒ Claim(s) *4-24* is/are pending in the application.

Of the above, claim(s) _____ is/are withdrawn from consideration.

☐ Claim(s) _____ is/are allowed.

☒ Claim(s) *4-7, 9-17, and 19-23* is/are rejected.

☒ Claim(s) *8, 18, and 24* is/are objected to.

☐ Claims _____ are subject to restriction or election requirement.

Application Papers

☐ See the attached Notice of Draftsperson's Patent Drawing Review, PTO-948.

☐ The drawing(s) filed on _____ is/are objected to by the Examiner.

☐ The proposed drawing correction, filed on _____ is ☐ approved ☐ disapproved.

☒ The specification is objected to by the Examiner.

☐ The oath or declaration is objected to by the Examiner.

Priority under 35 U.S.C. § 119

☐ Acknowledgement is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d).

☐ All ☐ Some* ☐ None of the CERTIFIED copies of the priority documents have been

☐ received.

☐ received in Application No. (Series Code/Serial Number) _____ .

☐ received in this national stage application from the International Bureau (PCT Rule 17.2(a)).

*Certified copies not received: _____

☐ Acknowledgement is made of a claim for domestic priority under 35 U.S.C. § 119(e).

Attachment(s)

☒ Notice of References Cited, PTO-892

☐ Information Disclosure Statement(s), PTO-1449, Paper No(s). _____

☐ Interview Summary, PTO-413

☐ Notice of Draftsperson's Patent Drawing Review, PTO-948

☐ Notice of Informal Patent Application, PTO-152

--- SEE OFFICE ACTION ON THE FOLLOWING PAGES ---

DETAILED ACTION

Response to Amendment

1. This action is a response to the Amendment filed on 10/17/96. Claims 1-3 have been canceled, and claims 4-24 are pending. The corrections to the Drawings are held in abeyance. The objection to the title is maintained for reasons set forth below.

Allowable Subject Matter

2. Claim 8 is objected to as being dependent upon a rejected base claim, but would be allowable if rewritten in independent form including all of the limitations of the base claim and any intervening claims.

3. The following is a statement of reasons for the indication of allowable subject matter: Claim 8 requires a specific equation for the mathematical model upon which the first bit allocation is based. The specific formula is not taught or suggested by the prior art of record. Claims 18 and 24 are dependent thereon, and therefore, are also objected to as being dependent upon a rejected base claim (5).

Response to Arguments

4. Applicant's arguments have been fully considered but they are not persuasive. Applicant argues (1) that Veldhuis et al. use set quantization levels for the lower half for frequency ranges and variable levels for the upper half, (2) that the instant application uses an allocation for all

frequency ranges, (3) that the instant application counts the number of available frequency units, and (4) claim 4 is not taught by Veldhuis et al. because "curve d [of Fig. 5] is not the same." (Amendment pp. 11).

As per arguments (1) and (2), the requirement of adaptive allocation to all frequencies is not specified in the claims, and is therefore irrelevant to consideration of prior art.

As per (3), Veldhuis et al. explicitly teach the use 26 separate frequency divisions based on psychoacoustic criterion which allow for wider variations of frequencies at higher frequency representations (Fig. 3, col. 'p' with corresponding frequency col. f_{CO}(Hz)). Applicant's specification explicitly defines the number of frequency units as 26 units divided along the frequency band using psychoacoustic criterion (Specification, pp. 10, line 2). Only rather than pre-specify the frequency ranges, applicants use a variance to define which will be assigned to "frequencies units." However, the result is the same, a frequency spectrum divided into 26 units which takes into account old and well known psychoacoustic criterion.

Therefore, regardless of whether one sequentially "assigns" frequencies based on psychoacoustic criterion, or simply defines the assignment in a predetermined table, the result is a frequency spectrum split into 26 units based on old and well known techniques readily available to one of ordinary skill in the art of speech encoding and psychoacoustic masking.

As per (4), "curve d" is clearly outside the generalized claim language requiring only a in-bank masking," and therefore irrelevant pending more precise claim language.

Therefore, Veldhuis et al. will be maintained in the rejection as set forth below.

Further, please note that the very redundant and somewhat confusing claim numbering/structure has produced a oddly numbered rejections. For clarity, the examiner has traversed the claims in order of structure and non-redundancy, i.e. claim dependency group (5, 4, 12, 17, and 23); followed by claim groups (6, 10, 15, 21), (7, 11, 16, 22), (8, 13, 18, 24), and (9, 14, 19, 20).

Original Claim	Identical Claims in Different Claim Dependencies
12	9-11, and 13
17	14-16, and 18
23	20-22 and 24

Claim Rejections - 35 USC § 112

5. Claims 12 (and 9-11, 13) are rejected under 35 U.S.C. 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which applicant regards as the invention.

As per claim 12 (and 9-11, 13), the requirement of each frequency-unit having a single spectral component is vague and confusing. As discussed above, and explicitly shown in Fig. 6 (instant Application) each frequency-unit has many spectral components through many frequencies totaling 26 units which must represent a complete spectrum of information. These units are then sometime represented or quantized (bit allocated) as an average of all the spectral components within the specified range, i.e. the total power or amplitude within a frequency unit. Or; the units could be represented as a single spectral component, i.e. a max. amplitude or power.

Regardless of which old and well known technique is used, the frequency-unit will have more than one spectral component, making the requirement indefinite.

Note, all other claims referring to the frequency unit use the proper terminology as discussed above, see claim 17 (and 14-16, 18) and claim 23 (and 20-22, 24).

Claim Rejections - 35 USC § 103

6. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

7. Claims 5-7, 9-12, 14-17, 19-23 are rejected under 35 U.S.C. 103(a) as being unpatentable over Veldhuis et al., (5,105,463) in view of Nishiguchi et al. (5,151,941).

As per claim 5 and 19, Veldhuis et al. teach a method for determining a bit allocation...comprising the steps of:

obtaining frequency unit intervals based on psychoacoustic criteria (Fig. 3 as discussed above);

and suggest adaptive bit allocation techniques including using a "math model" as described in paper 5, paragraph 6.

Further, Nishiguchi et al. teach a method for determining a bit allocation comprising a first and second bit allocation which incorporates psychoacoustic criteria (their initial allocation followed by a "correcting step" col. 2 line 64- col. 3 line 14; and psychoacoustics col. 4).

Therefore, it would have been obvious to one of ordinary skill in the art of adaptive bit allocation at the time of the invention to specify a two step process as taught by Nishiguchi et al. in the adaptive bit allocation taught by Veldhuis et al. because the number of available bits can change dynamically which results in bits being in surplus or deficit and leading to less efficient bit allocation, thus, requiring more bandwidth (Nishiguchi et al., col. 1).

As per claim 4, Veldhuis et al. teach the claim limitations as discussed in paper 5.

As per claims 12 (and 9-11), 17 (and 14-16) and 23 (and 20-22), Veldhuis et al. teach the frequency-unit being represented by the average and maximum amplitudes or power (G(p;m) a characteristic parameter that represents any of peak value or average or amplitude or power, col. 6 lines 28-35). Such determination depends only on the type of quantizer (line 35). Note, claims 13, 18, and 24 are identical, but are dependent upon claim 8.

As per claim 6, Veldhuis et al. teach the claim limitations as discussed in paper 5 (the previous rejection of claim 2).

As per claim 7, Veldhuis et al. teach the counted number of available frequency units (Instant specification defines this number as 26; spec. pp. 10, Veldhuis et al. show this number as 26, Fig. 3).

Conclusion

8. Applicant's amendment necessitated the new ground(s) of rejection presented in this Office action. Accordingly, **THIS ACTION IS MADE FINAL**. See MPEP § 706.07(a). Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for response to this final action is set to expire THREE MONTHS from the date of this action. In the event a first response is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event will the statutory period for response expire later than SIX MONTHS from the date of this final action.

9. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

Fielder et al. teach the use of a multi-pass bit allocation methodology (col. 6-col. 7, col. 22-col. 27.

Akagiri et al. teach the use of both temporal and frequency masking effects (abstract).

10. The examiner requests any subsequent amendments amending claims, include a complete copy of current, pending claims.

11. Any inquiry concerning this communication or earlier communications from the examiner should be directed to John Grover whose telephone number is (703) 305-3863.

JMG

John Michael Grover
Assistant Patent Examiner
December 27, 1996

ALLEN R. MACDONALD
SUPERVISORY PATENT EXAMINER
ART UNIT 2308

Notice of References Cited

Application No.	Applicant(s)		
08/161,798	Teh et al.		
Examiner		Group Art Unit	Page 1 of 1
John Michael Grover		2308	

U.S. PATENT DOCUMENTS

		DOCUMENT NO.	DATE	NAME	CLASS	SUBCLASS
	A	5,479,562	12/95	Fielder et al.	395	2.38
	B	5,260,980	11/93	AKAGIRI ET AL.	375	122
	C					
	D					
	E					
	F					
	G					
	H					
	I					
	J					
	K					
	L					
	M					

FOREIGN PATENT DOCUMENTS

		DOCUMENT NO.	DATE	COUNTRY	NAME	CLASS	SUBCLASS
	N						
	O						
	P						
	Q						
	R						
	S						
	T						

NON-PATENT DOCUMENTS

		DOCUMENT (Including Author, Title, Source, and Pertinent Pages)	DATE
	U		
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	W		
	X		

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PATENT
Amendment Under 37 CFR §1.116
Expedited Procedure
Examining Group 2308

10/B

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicant : TEH, et al.

Serial No.: 08/161,798 Examiner: Grover, J.

Filed : December 6, 1993 Group Art Unit: 2308

For : A BIT ALLOCATION METHOD FOR DIGITAL AUDIO SIGNALS (AS AMENDED)

AMENDMENT

Assistant Commissioner of Patents
and Trademarks
Washington, D.C. 20231

Sir:

Entry and consideration of this paper in response to the January 6, 1997 Office Action (Paper No. 8) are requested under 37 C.F.R. §1.116. Please further amend this application as follows:

please enter AM

IN THE CLAIMS:

Pursuant to the Examiner's request all pending claims are reproduced below including any claim amendments. Please amend claims 5 and 8-13 as follows:

4. Method according to claim 5, 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.

B1

5. (Amended) 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:

a step for dividing at least one frequency interval into a plurality of frequency-units;

a step for obtaining a representative of each frequency-unit obtained [by dividing at least one defined frequency interval];

a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria[, said available frequency-units satisfy said psychoacoustic criteria];

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

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

6. A method according to claim 5, wherein said second bit allocation step is to determine the bit allocation through a psychoacoustically weighted inverse relationship with the said representative of each of said frequency-units.

7. A method according to claim 5, wherein said counted number of available frequency-units is a number of the frequency-units which have the representative exceeding a human hearing threshold.

18. (Amended) A method [according to claim 7,] 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:

a step for dividing at least one frequency interval into a plurality of frequency-units;

a step for obtaining a representative of each frequency-unit obtained;

a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria where the representative exceeds a human hearing threshold;

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

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

wherein said mathematical model is represented by a formula:

$$WL(u) = \frac{var(u)}{F} - \frac{1}{N \cdot F} \sum_{u=0}^{N} var(u) + \frac{Ba}{A \cdot N}$$

wherein

WL(u) represents number of bits allocated to the frequency-units u,

var(u) represents variance of the frequency-units u,

F represents a psychoacoustic weighting factor,

N represents the counter number of the available frequency-units,

Ba represents number of bits set aside for initial allocation, and

A represents approximate average number of spectral samples in the available frequency-units.

9. (Amended) Method according to claim 5, wherein each frequency-unit [has] is represented by a single spectral component.

10. (Amended) Method according to claim 6, wherein each frequency-unit [has] is represented by a single spectral component.

11. (Amended) Method according to claim 7, wherein each frequency-unit [has] is represented by a single spectral component.

12. (Amended) Method according to claim 4, wherein each frequency-unit [has] is represented by a single spectral component.

13. (Amended) Method according to claim 8, wherein each frequency-unit [has] is represented by a single spectral component.

14. Method according to claim 5, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.

15. Method according to claim 6, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.

16. Method according to claim 7, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.

17. Method according to claim 4, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.

18. Method according to claim 8, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.

19. Method according to claim 5, wherein each frequency-unit is obtained by dividing said spectral and temporal structure with at least one frequency interval.

20. Method according to claim 5, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.

21. Method according to claim 6, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.

22. Method according to claim 7, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.

23. Method according to claim 4, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.

24. Method according to claim 8, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.

REMARKS

Reconsideration of the Office Action dated January 6, 1997 is respectfully requested under 37 C.F.R. §1.116 to better clarify the issues for appeal and to respond to the new art rejections raised by the Examiner.

Claims 4-24 are pending in this application. Claim 5 has been amended to more clearly define the inventive method. Claim 8 has been amended to include the features of claims 5 and 7 to be in independent form. Claims 9-13 have been amended in an attempt to overcome the 35 U.S.C. §112, second paragraph, rejection. No additional claim fees are required. No new matter or issues have been raised by the claim amendments. No amendments are made to the specification or Abstract of the Disclosure.

Applicants note that the Examiner is holding the drawing corrections proposed in the Amendment filed on October 17, 1996 in abeyance. Applicants also note that the objection to the title is maintained, but Applicants are unsure of the reasons for the maintenance as there were no reasons set forth in the Office Action of January 6, 1997 beyond the statement in paragraph 1.

Applicants note with appreciation that claim 8 was found allowable if rewritten in independent form. Applicants have amended claim 8 to include the features of claims 5 and 7 to satisfy this requirement. Claims 18 and 24 are believed to be

allowable as they now depend from allowable claim 8. It is also believed that claim 13 as amended to satisfy the requirements of 35 U.S.C. §112 is allowable also as it depends from claim 8.

Claims 9-13 have been rejected under 35 U.S.C. §112, second paragraph, for purportedly being indefinite. This rejection is respectfully traversed. Each of these claims has been amended to replace "has" with "is represented by" to now recite that each frequency unit is represented by a single spectral component. Applicants believe that this should satisfy the requirements of 35 U.S.C. §112 as discussed in paragraph 5 of the January 6, 1997 Office Action. It is believed that these amendments overcome this rejection, and it is respectfully requested that this rejection be withdrawn.

Claims 5-7, 9-12, 14-17, 19-23 are rejected under 35 U.S.C. §103(a) as purportedly unpatentable over Veldhuis et al. in view of Nishiguchi et al. This rejection is respectfully traversed.

Claim 5 is the base independent claim for the rejected claims. Claim 5 recites that at least one frequency interval is divided into a plurality of frequency units with no specific recitation of the number of frequency units that the frequency interval is divided into contrary to the Examiner's statements in the second full paragraph on page 3 of the January 6, 1997 Office Action. Applicants believe that although the specification may state that the frequency units can number 26, this does not place a specific limitation on the claim as recited. Claim 5 continues on to recite that obtaining a

representative for each frequency unit. The next step is to count the number of available frequency units determined based on using the representative of each frequency unit using psychoacoustic criteria to make the determination. Next, the method allocates bits for each of the available frequency units. The bit allocation occurs in two steps. The first step determines the initial bit allocation by using an approximate mathematical model that has as one of its factors the number of available frequency units. The second bit allocation step further fine tunes the bit allocation for each of the available frequency units based on psychoacoustic criteria.

The Examiner, as previously mentioned, apparently has mistaken and implied a limitation upon claim 5 based on the discussion within the specification. This implication of a limitation upon the recited claim language is improper and cannot be used as a basis for rejection. Applicants are confused by the Examiner's use of "variance" in the second full paragraph of page 3, as Applicants use variance in the specification to define a change each frequency unit. "Variance" is not used to assign or position the frequency units per se. The frequency units are the result of dividing a particular frequency interval into a plurality of frequency units. The Examiner is correct if he is using variance in relation to allowing for flexibility in the determination in the number of frequency units and their respective sizes. Once again, the number of frequency units is not specifically recited

and therefore there is no limitation in reference to their being exactly 26 frequency units.

The Examiner has improperly defined the step of obtaining frequency units by dividing a defined frequency interval as being based on psychoacoustic criteria. Psychoacoustic criteria is not used until the step of counting the number of available frequency units, which does not explicitly remove frequency units that are determined not to meet the psychoacoustic criteria for the bit allocation based on the representative. It is this representative that is based on the psychoacoustic criteria not the frequency units themselves per se. The Examiner relies upon column 3 of the Veldhuis et al. patent to teach a mathematical model being used for the allocation of bits, this is not entirely true as that particular reference refers to quantizing bits for a single block in response to an allocation pattern stored, which allocation pattern is B(p). *See* column 3, lines 4-8. The Examiner then goes on to rely upon the Nishiguchi et al. patent to allegedly teach a method for determining bit allocation comprising two steps which incorporate psychoacoustic criteria. The Examiner states that after the initial allocation there is a correcting step. This however is different than what is recited in Applicants' claim 5, which is that the second bit allocation step further allocates bits that were not previously allocated to each of the relevant available frequency units based on psychoacoustic criteria. This type of step is entirely different than a

correction step which by its nature does not allocate remaining bits but corrects the allocation previously made by adjusting bits previously allocated.

Furthermore, a combination of the Veldhuis et al. and Nishiguchi et al. patents is not proper. The Veldhuis et al. patent as mentioned in the first Amendment relies upon a quantinization of the lower half of the frequency ranges. *See* column 7, line 68 to column 8, line 1 and page 11 of the first Amendment. The Veldhuis et al. patent only disclose a bit allocation for the frequency ranges in the upper half of the frequency spectrum. *See* column 10, lines 8-18 and page 11 of the first Amendment. Consequently, the Veldhuis et al. patent teaches away from the use of bit allocation in the lower frequency spectrums. Consequently, if Nishiguchi et al. bit allocation is combined with the Veldhuis et al. patent, the lower frequency bands will still be quantized and not have bits variably assigned through allocation based on psychoacoustic data. Contrary to the Examiner's assertion that bit allocation does not possibly occur in the lower spectrum of frequencies of Applicants' invention, Applicants adjust the bit allocation for the available frequency units. The available frequency units can be and include any frequency unit whose representative satisfies psychoacoustic criteria. This does not explicitly limit the frequency units to only the upper half of the frequency spectrum of the selected frequency interval, however in a particular example, this may be the actual result.

However, the available frequency units may possibly include the lower frequency units in the lower half of the selected frequency interval. This possibility is not taught or suggested by either of the Veldhuis et al. patent or the Nishiguchi et al. patent.

Consequently, the deficiencies of each of the cited and relied upon patents of the Examiner are not remedied by the other respective patent. Consequently, the Veldhuis et al. patent and the Nishiguchi et al. patent, in combination or taken individually, are not able to teach or suggest Applicants' recited invention. Even though a few features were used to distinguish between this application and the relied upon art, it is the totality of the claims that provides the patentability for each claim. The remaining claims that depend from claim 5 are patentable for the same reasons as claim 5. Applicants hereby incorporate by reference the arguments made in the first Amendment filed on October 17, 1996. Therefore, this rejection has been overcome, and it is respectfully requested that this rejection be withdrawn.

In view of the above amendments and Remarks, it is courteously urged that all of the claims are allowable and that the application now is in condition for allowance. Favorable action in this regard is earnestly solicited.

Respectfully submitted,

BEVERIDGE, DEGRANDI, WEILACHER & YOUNG, L.L.P.

By: _____

Joseph A. Degrandi, Reg. No. 17,446
1850 M Street, N.W. (Suite 800)
Washington, D.C. 20036
Telephone: (202) 659-2811
Fax: (202) 659-1462

June 6, 1997

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03/58-48683



-90-117-4p.2308

#9 EXT 8/D Time 6/30/9

PATENT
Amendment Under 37 CFR §1.116
Expedited Procedure
Examining Group 2308

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicant : TEH, et al.

Serial No.: 08/161,798 Examiner: Grover, J.

Filed : December 6, 1993 Group Art Unit: 2308

For : A BIT ALLOCATION METHOD FOR DIGITAL AUDIO SIGNALS (AS AMENDED)

PETITION FOR EXTENSION OF TIME

Hon. Commissioner of Patents
and Trademarks
Washington, D.C. 20231

Sir:

Applicants, through their undersigned attorney, respectfully request a two-month extension of time for responding to the outstanding Office Action dated January 6, 1997 in the above-identified patent application. The time for response is thus extended up to and including June 6, 1997.

Our check in the sum of $390.00 is attached.

Please charge any additional fees to the Deposit Account of the undersigned, No. 02-4300, and credit any overpayment to said Account.

06/25/1997 AHAYES 00000003 08161798
01 FC:117 390.00 OP

Respectfully submitted,

Beveridge, DeGrandi, Weilacher & Young

By: Joseph A. DeGrandi

Joseph A. DeGrandi - Reg. No. 17,446
1850 M Street, N.W. (Suite 800)
Washington, D.C. 20036
Telephone: (202) 659-2811
Facsimile: (202) 659-1462

June 6, 1997

300/119

OMB No. 0651-0011 (12/31/86)

Notice of Appeal from the Examiner to the Board of Appeals and Interferences	Atty. Docket No. 03/58-48683
In Re Application of TEH, et al.	
Serial Number 08/161,798	Filed December 6, 1993
For: A BIT ALLOCATION METHOD FOR DIGITAL AUDIO SIGNALS (AS AMENDED)	
Grp. Art Unit 2308	Examiner

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Applicant hereby appeals to the Board of Appeals from the decision dated January 6, 1997 of the examiner rejecting claims 4-7, 9-17 and 19-23.

The item(s) checked below are appropriate:

1. X Appeal fee (37 CFR 1.17(e))

   X other than small entity — fee $300

   ☐ small entity — fee $150
   - ☐ verified statement attached
   - ☐ verified statement was filed

2. X A petition and fee for an extension of time for reply to the rejection is attached (the first and second month extension fees were previously paid on June 6, 1997).
   Extension fee (37 CFR 1.17(a)-(c))

   X other than small entity
   - ☐ $110 ☐ $390 ☐ $930 ☐ $1470
   - X $540 (third month only)

   ☐ small entity
   - ☐ verified statement attached
   - ☐ verified statement filed
   - ☐ $55 ☐ $195 ☐ $465 ☐ $735

3. X Check attached for the sum of $840.00.

4. ☐ Please charge the fee(s) to Deposit Account Number 02-4300. A duplicate copy of this notice is enclosed.

July 7, 1997
Date

Joseph A. DeGrandi
Applicant, Attorney or Agent
Joseph A. DeGrandi (17,446)

Form PTO-FB-B210 (10-85)
Patent and Trademark Office-U.S. Department of Commerce

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PATENT
Amendment Under 37 CFR §1.116
Expedited Procedure
Examining Group 2308

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IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicant : TEH, et al.

Serial No.: 08/161,798 Examiner:

Filed : December 6, 1993 Group Art Unit: 2308

For : A BIT ALLOCATION METHOD FOR DIGITAL AUDIO SIGNALS (AS AMENDED)

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GROUP 2300

PETITION FOR EXTENSION OF TIME

Hon. Commissioner of Patents
and Trademarks
Washington, D.C. 20231

Sir:

Applicants, through their undersigned attorney, respectfully request a third-month extension of time for responding to the outstanding Office Action dated January 6, 1997 in the above-identified patent application. The time for response is thus extended up to and including July 7, 1997, since July 6, 1997 was a Sunday.

Applicants previously submitted a Petition for Extension of Time for two months on June 6, 1997 along with a check in the amount of $390.

Our check including the sum of **$540.00** ($930.00-$390.00) is attached.

Please charge any additional fees to the Deposit Account of the undersigned, No. 02-4300, and credit any overpayment to said Account.

Respectfully submitted,

Beveridge, DeGrandi, Weilacher & Young

By: Joseph A. DeGrandi

Joseph A. DeGrandi - Reg. No. 17,446
1850 M Street, N.W. (Suite 700)
Washington, D.C. 20036
Telephone: (202) 659-2811
Facsimile: (202) 659-1462

July 7, 1997

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UNITED STATES DEPARTMENT OF COMMERCE
Patent and Trademark Office
Address: COMMISSIONER OF PATENTS AND TRADEMARKS
Washington, D.C. 20231

SERIAL NUMBER	FILING DATE	FIRST NAMED APPLICANT	ATTORNEY DOCKET NO.
08/161,798	12/06/93	TEH	D 0348683

26M1/0708
BEVERIDGE, DEGRANDI, WEILACHER & YOUNG
1850 M STREET, N. W., SUITE 800
WASHINGTON DC 20036

EXAMINER	
MACDONALD, A	
ART UNIT	PAPER NUMBER
2308	10

DATE MAILED: 07/08/97

Below is a communication from the EXAMINER in charge of this application

COMMISSIONER OF PATENTS AND TRADEMARKS

ADVISORY ACTION

☒ THE PERIOD FOR RESPONSE:

a) ☒ is extended to run 5 months or continues to run _____ from the date of the final rejection

b) ☐ expires three months from the date of the final rejection or as of the mailing date of this Advisory Action, whichever is later. In no event however, will the statutory period for the response expire later than six months from the date of the final rejection.

Any extension of time must be obtained by filing a petition under 37 CFR 1.136(a), the proposed response and the appropriate fee. The date on which the response, the petition , and the fee have been filed is the date of the response and also the date for the purposes of determining the period of extension and the corresponding amount of the fee. Any extension fee pursuant to 37 CFR 1.17 will be calculated from the date of the originally set shortened statutory period for response or as set forth in b) above.

☐ Appellant's Brief is due in accordance with 37 CFR 1.192(a).

☒ Applicant's response to the final rejection, filed 5-6-97 has been considered with the following effect, but it is not deemed to place the application in condition for allowance:

1. ☐ The proposed amendments to the claim and /or specification will not be entered and the final rejection stands because:

   a. ☐ There is no convincing showing under 37 CFR 1.116(b) why the proposed amendment is necessary and was not earlier presented.

   b. ☐ They raise new issues that would require further consideration and/or search. (See Note).

   c. ☐ They raise the issue of new matter. (See Note).

   d. ☐ They are not deemed to place the application in better form for appeal by materially reducing or simplifying the issues for appeal.

   e. ☐ They present additional claims without cancelling a corresponding number of finally rejected claims.

   NOTE: _____

2. ☐ Newly proposed or amended claims _____ would be allowed if submitted in a separately filed amendment cancelling the non-allowable claims.

3. ☒ Upon the filing an appeal, the proposed amendment ☒ will be entered ☐ will not be entered and the status of the claims will be as follows:

   Claims allowed: 8, 13, 18, 24
   Claims ~~objected to~~ Rejected: 4-7, 9-12, 14-17, 19-23
   Claims ~~rejected~~: _____

   However;

   ☐ Applicant's response has overcome the following rejection(s): _____

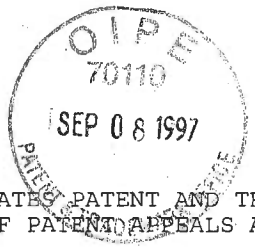
4. ☐ The affidavit, exhibit or request for reconsideration has been considered but does not overcome the rejection because _____

☐ The affidavit or exhibit will not be considered because applicant has not shown good and sufficent reasons why it was not earlier presented.

☐ ng correction ☐ has ☐ has not been approved by the examiner.

Albert Macdonald
MACDONALD

Atty. Dkt.
03/58-48683



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GP 2308
9-26-97

PATENT

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE
BEFORE THE BOARD OF PATENT APPEALS AND INTERFERENCES

Applicant	:	Do Hui TEH et al.	Examiner: A. MacDonald
Serial No.	:	08/161,798	Group Art Unit: 2308
Filed	:	December 6, 1993	
For	:	A BIT ALLOCATION METHOD FOR DIGITAL AUDIO SIGNALS (AS AMENDED)	

APPLICANTS' BRIEF ON APPEAL

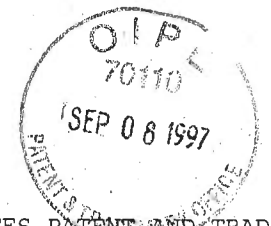
BEVERIDGE, DEGRANDI, WEILACHER & YOUNG, L.L.P
Joseph A. DeGrandi
PTO Reg. No. 17,446
1850 M Street, N.W., Suite 800
Washington, D.C. 20036
Telephone No.: (202) 659-2811
Attorney for Appellant

09/22/1997 JWASHING 00000184 08161798
01 FC:120
300.00 OP

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Filed	:	December 6, 1993	
For	:	A BIT ALLOCATION METHOD FOR DIGITAL AUDIO SIGNALS (AS AMENDED)	

APPELLANTS' BRIEF ON APPEAL

Applicants have appealed from the final Office Action dated January 6, 1997 (Paper No. 8) in the above application.

I. REAL PARTY IN INTEREST

This patent application is assigned to Matsushita Electric Industrial Co., Ltd., a corporation of Japan, and the assignment is recorded at Reel/Frame 6798/0252 with recordation taking place on December 6, 1993.

II. RELATED APPEALS AND INTERFERENCES

Applicants, Applicants' legal representative and Assignee know of no currently pending related appeals or interferences which will directly affect or be directly affected by or have a bearing on the Board's decision in the pending appeal.

III. STATUS OF CLAIMS

Claims 1-3 were previously canceled.

Claims 4-7, 9-12, 14-17, and 19-23 stand rejected after the final Office Action and the Advisory Action of July 8, 1997 (Paper No. 11).

Claims 8, 13, 18, and 24 were indicated as allowed in the Advisory Action of July 8, 1997, as amended in the Amendment filed June 6, 1997 (Paper No. 9) in response to the final Office Action. These claims are attached as Appendix B.

IV. STATUS OF AMENDMENTS

Applicants filed an Amendment on June 6, 1997 in response to the final Office Action of January 6, 1997. An Advisory Action issued in response to this Amendment on July 8, 1997 indicating that claims 8, 13, 18, and 24 were allowed. The Advisory Action stated that claims 4-7, 9-12, 14-17, and 19-23 stand rejected based on prior art. However, Applicants assume, based on the indication of allowance of claim 13, that the rejection based on 35 U.S.C. §112, second paragraph, was overcome also for claims 9-12 as these claims were also amended to recite the same feature as recited in claim 13, namely, that each frequency-unit is represented by a single spectral component. The Advisory Action indicated that the Amendment would be entered upon the filing of an Appeal. The Notice of Appeal was filed on July 7, 1997 (July 6 being a Sunday).

V. SUMMARY OF THE INVENTION

The invention, in the simplest terms, is a method of determining the amount of bit allocation for quantization of digital audio signals. The digital audio signals have spectral and temporal structure and are obtained by buffering audio signals in frames and decomposing the digital signals into spectral components.

The first recited step is to divide a frequency interval into a plurality of frequency-units. For each frequency-unit, a representative is obtained. *See* the specification at page 5, lines 10-13. The representative then is used in conjunction with psychoacoustic criteria to count and determine a number of available frequency-units. *See* the specification at page 10, lines 7-11 and Figure 2, boxes 32-35. A first bit allocation step determines the initial bit allocation for each of the available frequency-units by using a mathematical model. *See* the specification at page 11, line 3 to page 12, line 1. A second bit allocation step then further determines bit allocation for each of the available frequency-units based on psychoacoustic criteria. *See* the specification at page 9, lines 9-12.

The mathematical model may be derived from an in-band masking affect that allows an error spectrum to more closely follow the original spectrum. *See* the specification at page 12, lines 19-21. The counted number of available frequency-units may be based on frequency-units that have a representative exceeding

a human hearing threshold. *See* the specification at page 10, lines 7-11. Other possible further modifications are that each frequency-unit is represented by a single spectral component. The representative may be a variance which is calculated from the amplitude of the maximum spectral component of each frequency-unit. *See* the specification at page 8, lines 7-9. The representative also may be a variance which is calculated from an average of a spectral component of each frequency-unit. *See* the specification at page 8, lines 9-11.

VI. ISSUES

1. The only issue before the Board is whether claims 5-7, 9-10, 13-17, and 19-23 define a patentable invention under 35 U.S.C. §103(a) over the Veldhuis et al. patent No. 5,105,463 in view of the Nishiguchi et al. patent No. 5,151,941.

VII. GROUPING OF CLAIMS

Claims 4-17, 9-12, 14-17, and 19-23 stand or fall together as a single group of claims.

VIII. ARGUMENT

The Examiner is in error in rejecting claims 5-7, 9-12, 14-17, and 19-23 under 35 U.S.C. §103(a) as allegedly being

unpatentable over the Veldhuis et al. patent in view of the Nishiguchi et al. patent.

The general subject matter of this invention is the digital manipulation of an audio signal based on the principles of psychoacoustics. Psychoacoustics relates to how an individual perceives a sound based on temporal, spectral, and spacial factors. Psychoacoustics accounts for the way soundwaves reverberate off of the ridges and folds of an individual's ear. As such, certain frequency ranges become more important than others and therefore do not need to be as finely represented in the processing. This does not mean, however, that all frequency bands available in a particular range are not processed or utilized. Furthermore, there is no explicit or mandatory number of bands that must be used to accomplish this processing.

Claim 5 recites that psychoacoustic criteria is used for determining the number of available frequency-units based on the representative assigned to each frequency-unit. There is no specific recited limitation that prevents each of the frequency-units from being an available frequency-unit. Furthermore, the specification recites that the steps recited in claim 5 are reiterated until the bit allocation of all of the bits has occurred. Consequently, one of ordinary skill in the art would understand this as meaning that each frequency-unit would be an available frequency-unit on the first iteration. However, as iterations are performed, the representative is adjusted and therefore may not satisfy the psychoacoustic criteria, thus, the

frequency-unit would not be an available frequency-unit. The consequence of not being an available frequency-unit is that no additional bits are assigned or allocated to that frequency-unit.

Applicants do not understand the interpretation given to the claims by the Examiner in the final Office Action. As will be discussed below, Applicants respectfully submit that the invention recited in claim 5 is patentable over the relied upon Veldhuis et al. patent and the Nishiguchi et al. patent.

The Examiner has unnecessarily read limitations into claim 5 that are not recited. In paragraph 4 of the final Office Action, the Examiner implies that there is a limitation within the claims that the number of frequency-units is equal to 26. The Examiner states that:

> Applicant's specification explicitly defines the number of frequency units as 26 units divided along the frequency band using psychoacoustic criterion (Specification, pp. 10, line 2). Only rather than pre-specify the frequency ranges, applicants use a variance to define which will be assigned to "frequencies units." However, the result is the same, a frequency spectrum divided into 26 units which takes into account old and well known psychoacoustic criterion.

The problem with the Examiner's statement is that the number "26" is used as an example within Applicants' specification, and is not recited in Applicants' claim 5. Additionally, the Examiner, like Applicants, should not read a limitation from the specification into the claims. The *Manual of Patent Examining Procedure*, on page 2100-126, explicitly states that "[a]lthough

the claims are interpreted in light of the specification, limitations from the specification are not read into the claims. *In re Van Geuns*, 988 F.2d 1181, 26 USPQ2d 1057 (Fed. Cir. 1993)." Consequently, the number of frequency-units can be any number. Furthermore, the Examiner has limited the use of the representative within claim 5 by referring to it as a variance; however, a variance is a feature recited in the dependent claims as to what the representative may be.

Claim 5 is the independent claim from which all of the remaining rejected claims depend ultimately. Claim 5 recites that at least one frequency interval is divided into a plurality of frequency-units with no specific recitation of the number of frequency-units that the frequency interval is divided into, contrary to the Examiner's statements in the second full paragraph on page 3 of the January 6, 1997 Office Action, namely the above quoted text. Applicants believe that although the specification may state that the number of frequency-units can be 26, this does not place a specific limitation on claim 5 as recited.

Claim 5 also recites the step of "obtaining a representative for each frequency-unit obtained" and the step of counting the number of available frequency-units determined by using the representative of each frequency-unit, using psychoacoustic criteria to make the determination. Next, the method defined by claim 5 allocates bits for each of the available frequency-units. This bit allocation occurs in two steps. The first step

determines the initial bit allocation by using an approximate mathematical model (it should be noted that allowed claim 8 has a specific mathematical model in the form of an equation) that has as one of its factors the number of available frequency-units. The second bit allocation step further fine-tunes the bit allocation to each of the available frequency-units based on psychoacoustic criteria.

The Examiner, as previously mentioned, apparently has mistaken and implied a limitation upon claim 5 based on the discussion within the specification. This implication of a limitation upon the recited claim language is improper and cannot be used as a basis for rejection. Applicants are confused by the Examiner's use of "variance" in the second full paragraph of page 3, when he states "[o]nly rather than pre-specify the frequency ranges, applicants use a variance to define which will be assigned to 'frequency units.'"

Applicants use the term "variance" in the specification to define a change within each frequency-unit. "Variance" is not used to assign or position the frequency-units *per se*. The frequency-units are the result of dividing a particular frequency interval into a plurality of frequency-units. Once again, the number of frequency-units is not specifically recited in claim 5 and therefore there is no limitation in the claim reference to there being exactly 26 frequency-units.

The Examiner has implicitly admitted that the Veldhuis et al. patent does not teach, by itself, Applicants' recited invention by his withdrawal of the 35 U.S.C. §102 rejection.

The Veldhuis et al. patent discloses the use of multiple frequency bands as shown in Fig. 3. The Veldhuis et al. patent, however, does not determine a representative for each of these frequency bands. Instead, the Veldhuis et al. patent relies upon using a set number of quantinization levels that are predetermined for the lower half of the frequency ranges. As disclosed in the following:

> The quantizers 8(p) and dequantizers 9(p) for each of the subbands of the first group ($1 \leq p \leq 12$) are arranged for quantizing and dequantizing the subband signals by a fixed number of B(p) bits per signal sample, in the present embodiment the same values of $B(p=\log_2[L(p)]$ as shown in the table of FIG. 3 being chosen, thus L(p)=25 and B(p)=4.64. (Veldhuis et al., col. 10, lines 1-7).

L(p) represents the number of quantinization levels assigned for a given frequency band. The Veldhuis et al. patent does discuss the alloation for the upper half of the frequency ranges. *See* Veldhuis et al., col. 10, lines 8-18.

This automatic assignment of quantinization levels dictates a fixed amount of bits being assigned to the lower frequency bands. Consequently, the Veldhuis et al. patent teaches away from a step of "counting a number of available frequency-units by using the representative, based on psychoacoustic criteria." *See* Applicants' claim 5. The Examiner states on page 3 of the Final Office Action that "regardless of whether one sequentially

'assigns' frequencies based on psychoacoustic criterion, or simply defines the assignment in a predetermined table." This would indicate that the Examiner realizes that there is a difference between the use of psychoacoustic criterion and the use of preassigned relevant frequency bands as discussed in the Veldhuis et al. patent. Consequently, the Veldhuis et al. patent teaches only an apparatus that has predetermined characteristics and does not provide flexibility to the extent that is provided by Applicants' recited invention. Applicants allow for the possibility of assigning and allocating bits to the whole frequency spectrum, depending on where the relevant frequency-units are located based on the representative in comparison to psychoacoustic criteria. The Veldhuis et al. patent discloses a determination that the lower frequency bands do not require a flexible allocation of bits. *See* Veldhuis et al., col. 9, lines 53-67.

The Examiner has relied upon the Veldhuis et al. patent to also allegedly suggest adaptive bit allocation techniques using a mathematical model. Applicants' review of the cited portion of the Veldhuis et al. patent, column 3, lines 3-29, does not indicate the presence of a mathematical algorithm. At best, there is a listing of various variables that represent the number of bits to be allocated, characteristic parameters, and comparator signals. There is no teaching or suggestion within the Veldhuis et al. patent on how these variables would be construed as a mathematical model beyond contingent comparison

analysis as discussed at col. 11, lines 22-31. Furthermore, the Examiner admits that the Veldhuis et al. patent does not disclose two separate bit allocation steps by his reliance on the Nishiguchi et al. patent.

The Examiner relies upon the Nishiguchi et al. patent as allegedly teaching a method for determining a bit allocation comprising a first and second bit allocation steps, which each incorporates psychoacoustic criteria. The Examiner states that after the initial allocation there is a correcting step and refers to col. 2, line 64 to col. 3, line 14, and psychoacoustics, col. 4. This method, however, is different than what is recited in Applicants' claim 5, which is that the second bit allocation step further allocates bits that were not previously allocated to each of the relevant available frequency-units based on psychoacoustic criteria. This step is entirely different than a correction step which by its nature does not allocate remaining bits but corrects the allocation previously made by adjusting bits previously allocated. The steps disclosed by the Nishiguchi et al. patent do not teach or suggest an allocation based on the number of available frequency-units as recited in Applicants' claim 5.

Furthermore, a combination of the Veldhuis et al. and Nishiguchi et al. patents is not proper. The Veldhuis et al. patent relies upon a fixed quantization of the lower half of the frequency ranges. *See* Veldhuis et al., col. 7, line 68 to col. 8, line 1. The Veldhuis et al. patent only discloses a bit

allocation for the frequency ranges in the upper half of the frequency spectrum. *See* Veldhuis et al., col. 10, lines 8-18. Consequently, the Veldhuis et al. patent teaches away from the use of bit allocation in the lower frequency spectrums especially in light of the motivation alleged by the Examiner to combine these two patents. Consequently, if the Nishiguchi et al. bit allocation is combined with the Veldhuis et al. disclosure, the lower frequency bands will still be quantized and not have bits variably assigned through allocation based on psychoacoustic data. However, if bits are allocated to the full frequency spectrum, then the purpose of the Veldhuis et al. invention is destroyed.

Contrary to the Examiner's assertion that bit allocation does not possibly occur in the lower spectrum of frequencies of Applicants' invention, Applicants adjust the bit allocation for the available frequency-units. The available frequency-units can be and include any frequency-unit whose representative satisfies psychoacoustic criteria. This does not explicitly limit the frequency-units to only the upper half of the frequency spectrum of the selected frequency interval. However, the available frequency-units may possibly include the lower frequency-units in the lower half of the selected frequency interval. This possibility is not taught or suggested by either of the Veldhuis et al. patent or the Nishiguchi et al. patent.

Consequently, the deficiencies of each of the cited and relied upon patents of the Examiner are not remedied by the other

respective patent. Consequently, the Veldhuis et al. patent and the Nishiguchi et al. patent, in combination or taken individually, are not able to reach of suggest Applicants' recited invention. The remaining claims that depend from claim 5 are patentable for the same reasons as claim 5. Therefore, this rejection is in error, and these claims should be deemed patentable over the relied upon art.

Furthermore, claim 5 is similar to allowed claim 8 with the difference being that claim 8 includes the use of psychocoustic criteria to count the number of variable frequency-units based on the human hearing threshhold, and an exact mathematical model as represented by an express formula. It is Applicants' contention that the presence of the mathematical model in a general form provides patentability over the combination of the Veldhuis et al. patent and the Nishiguchi et al. patent, as neither of these patents teach or suggest the use of an approximate mathematical model that relies upon the number of counted available frequency-units as recited in claims 5 and 8. One of ordinary skill in the art would understand what is required by an approximate mathematical model. For this additional reason claim 5 and the claims that depend from it are patentable over the Veldhuis et al. patent and the Nishiguchi et al. patent, either in combination or individually.

IX. CONCLUSION

For the above reasons discussed in the Argument section, Applicants' claimed invention is not taught or suggested by the Veldhuis et al. and Nishiguchi et al. patents. All of the specific limitations in the claims are not described in the prior art relied upon in this rejection. Furthermore, the combination of these two patents does not teach or suggest the recited invention. Also, all of the features relied upon by the Examiner may not be properly combined. Therefore, the Examiner made an improper rejection of Applicants' recited invention based on a combination of the Veldhuis et al. and Nishiguchi et al. patents. Therefore, the claims of Applicants' invention are patentable.

Reversal of the final rejection is earnestly solicited.

Respectfully submitted,

BEVERIDGE, DEGRANDI, WEILACHER & YOUNG, L.L.P

By: Joseph A DeGrandi

Joseph A. DeGrandi
PTO Reg. No. 17,446
1850 M Street, N.W., Suite 800
Washington, D.C. 20036
Telephone No.: (202) 659-2811

Date: September 8, 1997

APPENDIX A

4. Method according to claim 5, 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.

5. 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:

a step for dividing at least one frequency interval into a plurality of frequency-units;

a step for obtaining a representative of each frequency-unit obtained;

a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria;

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

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

6. A method according to claim 5, wherein said second bit allocation step is to determine the bit allocation through a psychoacoustically weighted inverse relationship with the said representative of each of said frequency-units.

7. A method according to claim 5, wherein said counted number of available frequency-units is a number of the frequency-units which have the representative exceeding a human hearing threshold.

9. Method according to claim 5, wherein each frequency-unit is represented by a single spectral component.

10. Method according to claim 6, wherein each frequency-unit is represented by a single spectral component.

11. Method according to claim 7, wherein each frequency-unit is represented by a single spectral component.

12. Method according to claim 4, wherein each frequency-unit is represented by a single spectral component.

14. Method according to claim 5, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.

15. Method according to claim 6, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.

16. Method according to claim 7, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.

17. Method according to claim 4, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.

19. Method according to claim 5, wherein each frequency-unit is obtained by dividing said spectral and temporal structure with at least one frequency interval.

20. Method according to claim 5, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.

21. Method according to claim 6, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.

22. Method according to claim 7, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.

23. Method according to claim 4, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.

APPENDIX B

8. A method of determining a bit allocation for a quantization of digital audio signals having spectral and temporal structure w1

herein 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:

a step for dividing at least one frequency interval into a plurality of frequency-units;

a step for obtaining a representative of each frequency-unit obtained;

a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria where the representative exceeds a human hearing threshold;

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

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

wherein said mathematical model is represented by a formula:

$$WL(u) = \frac{var(u)}{F} - \frac{1}{N \cdot F} \sum_{u=0}^{N} var(u) + \frac{Ba}{A \cdot N}$$

wherein

WL(u)	represents number of bits allocated to the frequency-units u,
var(u)	represents variance of the frequency-units u,
F	represents a psychoacoustic weighting factor,
N	represents the counter number of the available frequency-units,
Ba	represents number of bits set aside for initial allocation, and
A	represents approximate average number of spectral samples in the available frequency-units.

13. Method according to claim 8, wherein each frequency-unit is represented by a single spectral component.

18. Method according to claim 8, wherein said representative is a variance which is calculated from an amplitude of a maximum spectral component of each frequency-unit.

24. Method according to claim 8, wherein said representative is a variance which is calculated from an average of a spectral component of each frequency-unit.

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BEFORE THE BOARD OF PATENT APPEALS AND INTERFERENCES

Paper No. 15

Application Number: 08/161,798

Filing Date: 12/06/93

Appellant(s): Teh et al

Joseph A. Degrandi

For Appellant

EXAMINER'S ANSWER

This is in response to appellant's brief on appeal filed September 26, 1997.

(1) Real Party in Interest

A statement identifying the real party in interest is contained in the brief.

(2) Related Appeals and Interferences

A statement identifying the related appeals and interferences which will directly affect or be directly affected by or have a bearing on the decision in the pending appeal is contained in the brief.

(3) Status of Claims

The statement of the status of the claims contained in the brief is correct.

(4) Status of Amendments After Final

The appellant's statement of the status of amendments after final rejection contained in the brief is correct.

(5) Summary of Invention

The summary of invention contained in the brief is correct.

(6) Issues

The appellant's statement of the issues in the brief is correct.

(7) Grouping of Claims

The appellant's statement in the brief that certain claims do not stand or fall together is not agreed with because a grouping of claims contain claims that have been allowed. In particular, appellant state that "claims 4-17, 9-12, 14-17, and 19-23 stand or fall together as a single group of claims". Examiner disagrees on the grouping of "claims 4-17". The only issue before the Board of Appeals is whether claims 5-7, 9-10, 13-17, and 19-23 define a patentable invention under 35 U.S.C. 103(a) over the Veldhuis et al (5105463) in view of Nishiguchi et al (5151941), as stated under Section VI of the Appeal Brief. Therefore, the grouping of claims should read "Claims 4-7, 9-12, 14-17, and 19-23 stand or fall together as a single group of claims".

(8) Claims Appealed

The copy of the appealed claims contained in the Appendix to the brief is correct.

(9) Prior Art of Record

The following is a listing of the prior art of record relied upon in the rejection of claims under appeal.

5105463	Veldhuis et al	4-1992
5151941	Nishiguchi et al	9-1992

(10) New Prior Art

No new prior art has been applied in this examiner's answer.

(11) Grounds of Rejection

The following ground(s) of rejection are applicable to the appealed claims:

Claims 5-7, 9-12, 14-17, 19-23 are rejected under 35 U.S.C. 103. This rejection is set forth in prior Office action, Paper No. 8, page 5, paragraph #7.

(13) Response to argument

Appellants' arguments filed 9/26/97 have been fully considered, but they are not persuasive.

Appellant argues against the art rejection of Veldhuis in view of Nishiguchi as applied to claim 5.

On pages 5-6, appellant discussed general subject matter and merely recites claim language. On page 7, applicant's argue that "Examiner implies that there is a limitation within the claims" (Appeal Brief, page 7 lines 11-12) and that "the Examiner should not read a limitation from the specification into the claims" (Appeal Brief, page 7, lines 18-19). Furthermore, applicant argues "the Examiner has limited the use of the representative within claim 5 by referring to it as a variance; however, a variance is a feature recited in the dependent claims as to what the representative may be" (Appeal Brief, page 8 lines 5-8). Also, applicant continues this argument (Appeal Brief, bottom of page 7 to page 9, line 12). Examiner rebuts with the fact that the argument presented (in paragraph 4 of the Final Office Action, paper #8) by the examiner attempts to clearly show the applicant that the technique used by the applicant is old and well known (See office action # 8, page 3, 2nd full paragraph). Furthermore, examiner argues that an example of prior art (office action # 8, page 3, 2nd full paragraph) that teaches a narrow limitation (the "variance", which is a feature in the dependent claim - Appeal Brief, page 8 lines 7-8) reads on the broader based claim (claim 5 - "the use of the representative" - Appeal Brief, page 8, lines 5-6).

On page 9 line 12 - bottom of page 9 (Appeal Brief), applicant's argue that "'Variance' is not used to assign or position the frequency units per se." in the Applicant's specification.

Examiner disagrees and points to Applicant's specification (page 10, lines 17-20), where "if the variance is above the threshold, the bandwidth counter is incremented". Therefore, the variance is used to assign the bandwidth (i.e., frequency units).

On page 10, applicant argues that Veldhuis et al teaches: 1) a set number of quantization levels that are predetermined for the lower half of the frequency ranges, 2) automatic assignment of quantization levels, 3)an apparatus that has predetermined characteristics and does not provide flexibility to the extent that is provided by Applicant's recited invention, and 4) a determination that the lower frequency bands do not require a flexible allocation of bits. Examiner argues that Veldhuis does teach adaptive bit allocation (col. 2 line 40 - col. 3 line 14). Examiner does agree that Veldhuis et al teaches fixed quantization levels, however, one cannot show non-obviousness by attacking references individually where the rejections are based on combinations of references. *In re Keller*, 642 F.2d 413, 208 USPQ 871 (CCPA 1981); *In re Merck & Co., Inc.*, 800 F.2d 1091, 231 USPQ 375 (Fed. Cir. 1986).

Applicant argues on page 11 and pages 14 (lines 7-16), that Veldhuis does not teach adaptive bit allocation techniques using a mathematical model. Examiner disagrees and points to Veldhuis et al(col. 2 lines 54 - 64) showing B(p), G(p;m) and C(p;m). Examiner notes that

a mathematical model or algorithm is a series of mathematical steps, as is shown in Veldhuis et al in the calculation of these parameters (Veldhuis et al, Fig. 8, Fig. 9).

Applicant argues that although Nishiguchi et al teaches a first and second bit allocation step (Appeal brief -page 12, lines 5-21), the second bit allocation step being a correcting step, is different "from what is recited in Applicant's claim 5, which is that the second bit allocation step further allocated bits that were not previously allocated to each of the relevant available frequency units based on psychoacoustic criteria" (appeal brief, page 12, lines 11-15). Examiner disagrees and argues that the claim language of claim 5 states "a second bit allocation step for determining further bit allocation for each of said available frequency units based on psychoacoustic criteria" (Amendment B, paper #10, page 2 lines 18-20). Nishiguchi teaches corrective bit allocation step (Nishiguchi, col. 3 lines 7-10 allocating bits) for each band (Nishiguchi col. 3 lines 1-5 - the difference between the energy of a band and the output of the noise level setting is calculated for each band, and then corrected), based on psychoacoustic criteria (Nishiguchi, col. 4 line 23 - human auditory characteristics). Furthermore, one cannot show non-obviousness by attacking references individually where the rejections are based on combinations of references. *In re Keller*, 642 F.2d 413, 208 USPQ 871 (CCPA 1981); *In re Merck & Co., Inc.*, 800 F.2d 1091, 231 USPQ 375 (Fed. Cir. 1986).

Appellant's argue that a combination of Veldhuis et al and Nishiguchi et al is not proper because Veldhuis teaches away from the use of bit allocation in the lower frequency spectrums and, if the bits are allocated to the full frequency spectrum because of the combination of Veldhuis in view of Nishiguchi et al, then the purpose of Veldhuis et al inventions is destroyed (Appeal Brief, page 12, line 22 - page 13, line 12). Examiner disagrees in that Nishiguchi teaches adaptive bit allocation of all frequency bands (Nishiguchi et al, col. 3 lines 1-10), and that the purpose of the Veldhuis et al invention would not be destroyed in combination with the Nishiguchi et al reference because the two step bit allocation process as taught by Nishiguchi et al leads to a more efficient bit allocation process. (See Veldhuis et al, col. 2 lines 22-24 -- their concern of not increasing the number of bits per sample; Nishiguchi is teaching a technique that would further reduce the number of bits used (i.e., increased bit efficiency, Nishiguchi col. 1)

Applicant argues that Examiner asserts that bit allocation does not possibly occur in the lower spectrum of frequencies of Applicants' invention (Appeal Brief, page 13, line 13-15), examiner disagrees and points to Office Action #8 page 3, first full paragraph, wherein examiner is addressing the fact that "the requirement of adaptive allocation to all frequencies is not specified in the claims". Furthermore, applicant argues that neither Veldhuis nor Nishiguchi teach this. Examiner disagrees and points to Nishiguchi (col. 3 lines 1-14) teaching adaption bit allocation for all bands. Furthermore, one cannot show non-obviousness

by attacking references individually where the rejections are based on combinations of references. *In re Keller*, 642 F.2d 413, 208 USPQ 871 (CCPA 1981); *In re Merck & Co., Inc.*, 800 F.2d 1091, 231 USPQ 375 (Fed. Cir. 1986).

For the above reasons, it is believed that the rejections should be sustained.

Respectfully submitted,

Michael N. Opsasnick

12/04/97

DAVID R. HUDSPETH
PRIMARY EXAMINER
GROUP ~~2500~~ 2741

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03/58-48683

PATENT

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE
BEFORE THE BOARD OF PATENT APPEALS AND INTERFERENCES

Applicant	:	Do Hui TEH et al.	Examiner: M. Opsasnick
Serial No.	:	08/161,798	Group Art Unit: 2741
Filed	:	December 6, 1993	
For	:	A BIT ALLOCATION METHOD FOR DIGITAL AUDIO SIGNALS (AS AMENDED)	

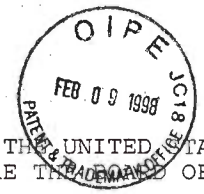
APPLICANTS' REPLY BRIEF

BEVERIDGE, DEGRANDI, WEILACHER & YOUNG, L.L.P.
Joseph A. DeGrandi
PTO Reg. No. 17,446
1850 M Street, N.W., Suite 800
Washington, D.C. 20036
Telephone No.: (202) 659-2811
Attorney for Appellant

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REPLY BRIEF TO EXAMINER'S ANSWER UNDER 37 C.F.R. §1.193(b)(1)

This Reply Brief is submitted in response to the Examiner's Answer dated December 9, 1997 in the above-identified application, in accordance with 37 C.F.R. §1.193(b)(1). Applicants submit this Reply Brief in order to clarify points of argument which have been misunderstood by the Examiner and set forth in the Examiner's Answer. This Reply Brief is also made as a matter of right under 37 C.F.R. §1.193(b)(1) to respond to the Examiner's Answer.

Applicants note from the Examiner's Answer that a new Examiner has been assigned to this application. The new Examiner has dropped the prior argument based on 26 frequency-units being taken from the specification and inserted into the claim language and being taught by the Veldhuis et al. patent. Applicants also note that the new Examiner has clarified the confusing basis for an argument related to "variance" made by a prior Examiner. Further, Applicants note that the new Examiner, at page 3, Section 7, has correctly stated the grouping of claims as claims 4-7, 9-12, 14-17, and 19-23.

Applicants believe that the present Examiner is in error in maintaining the rejection of claims 4-7, 9-12, 14-17, and 19-23 under 35 U.S.C. §103(a) as allegedly being unpatentable over the Veldhuis et al. patent in view of the Nishiguchi et al. patent. Furthermore, Applicants respectfully submit that the Examiner has yet to establish a *prima facie* obviousness rejection of these claims.

> To establish *prima facie* obviousness of a claimed invention, all the claim limitations must be taught or suggested by the prior art. *In re Royka*, 490 F.2d 981, 180 USPQ 580 (CCPA 1974). 'All words in a claim must be considered in judging the patentability of that claim against the prior art.' *In re Wilson*, 424 F.2d 1382, 165 USPQ 494, 496 (CCPA 1970). If an independent claim is nonobvious under 35 U.S.C. 103, then any claim depending therefrom is nonobvious. *In re Fine*, 837 F.2d 1071, 5 USPQ2d 1596 (Fed. Cir. 1988).

MPEP 2143.03 (Rev. 3, July 1997, p. 2100-112).

It appears that the last four Examiners[1] to review this case have on the whole adopted the rationale of the first Examiner in rejecting independent claim 5, based on art that does not teach or suggest each and every feature recited in claim 5. An example of this occurs in the Examiner's Answer at pages 6-7, where the current Examiner alleges that the Veldhuis et al. patent teaches adaptive bit allocation using a mathematical model

[1]The first Examiner wrote the first Office Action of June 17, 1996, rejecting the claims solely on the Veldhuis et al. patent. The second Examiner wrote the second Office Action of January 6, 1997, rejecting claims 4-7, 9-12, 14-17, and 19-23 on a combination of the Veldhuis et al. and Nishiguchi et al. patents. The third Examiner (the Supervisory Patent Examiner for former Art Unit 2308) signed the two Office Actions and the July 9, 1997 Advisory Action. The fourth Examiner wrote the Examiner's Answer, which dropped at least one prior argument. The fifth Examiner signed the Examiner's Answer.

or algorithm. Claim 5 recites "a first bit allocation step . . . using an approximate mathematical model using said counted number of available frequency-units." (emphasis added). Applicants were directing the Examiner's attention to the underlined portion of the claim and that the Veldhuis et al. patent does not teach a mathematical model based on the number of available frequency-units.[2]

The Examiner has correctly excerpted a portion of *In re Keller*, 642 F.2d 413, 208 USPQ 871 (CCPA 1981) as stated in the MPEP; however, the *Keller* decision is not relevant to the argument made in the present application. Keller used an expert who correctly stated that a published article did not teach the presence of a feature, "A", in combination with a second feature, "B".[3] The article was used in combination with two patents which taught the second feature, "B", thus both features, "A" and "B", were taught in the combination. *Id.* at 881. In the present application, the references relied upon by the Examiner do not disclose either feature "A" or feature "B". If a first reference relied upon for a particular feature, "A", by the Examiner does not actually have that feature, "A", as set forth in the claims, then a combination using that first reference with a second reference that also does not teach that feature, "A", would not

[2]*See* Applicants' Appeal Brief page 14, lines 8-18.

[3]The affidavit from the expert was characterized by the appellant as concerning "itself mainly with the question of whether the Walsh et al. article suggest [sic] the use of digital timing in a cardiac pacer." *Id.* at 879. Feature "A" would be the digital time and feature "B" would be the cardiac pacer.

remedy the defect (absence of feature "A") of the first reference. The *Keller* case did not involve the present fact situation and thus has no bearing on this application.

Furthermore, the Examiner has unnecessarily turned to the specification even through the language of claim 5 is clear as to its meaning. The Examiner, at page 5 of the Examiner's Answer, has taken the term "variance" from claims depending from claim 5[4] and taken an excerpt from page 10 of the specification describing the use of "variance" in rejecting claim 5 even though "variance" is not present in claim 5. By performing the analysis in this manner, the Examiner has modified the clear language in claim 5 in an effort to support his rejection based on the prior art.

Claim 5 recites a method of determining a bit allocation comprising five discrete steps, each with its own set of limitations. The Examiner is relying upon two patents, the first to Veldhuis et al. and the second to Nishiguchi et al., as allegedly teaching each of the five steps. The relied upon portions of these two patents taken in combination do not teach each and every limitation of claim 5. Accordingly, the Examiner is ignoring the clear language used in claim 5 in order to argue that the invention clearly defined in claim 5 is obvious from the teachings of the two patents, based on the Examiner's erroneous modification and interpretation of claim 5.

[4]*See* claims 14-17 and 20-23.

The first step recited in claim 5 is "dividing at least one frequency interval into a plurality of frequency-units." With respect to this step, Applicants submit that prior Examiners unnecessarily read additional limitations into this step by stating explicitly that there were 26 frequency-units. From the Examiner's Answer, Applicants believe that the current Examiner agrees that this was error on the part of the prior Examiners.[5]

The second step recited in claim 5 is to obtain "a representative of each frequency-unit" created in the first step. The Examiner on page 6 has quoted Applicants' specification by stating "where 'if the variance is above the threshold, the bandwith counter is incremented'. Therefore, the variance is used to assign the bandwidth (i.e., frequency-units)." Applicants respectfully submit that there is no mention of a

[5]There is no mention in the Examiner's Answer of the 26 frequency-units in the portion responding to the specification argument, but only discussion of the term "variance". Examiner's Answer, page 5, line 4 to page 6, line 3. However, the current Examiner appears to be under the mistaken belief that the representative (or variance in dependent claims) is used to subdivide a frequency interval into frequency-units. *Id.* This understanding of Applicants is based on the Examiner's use of the term "bandwith", which is defined as "*[t]elecommunications.* the smallest range of frequencies constituting a band, within which a particular signal can be transmitted without distortion." *Random House Unabridged Dictionary*, page 163 (2nd ed.) (emphasis in original). The term "band" is "[a]lso called **frequency band, wave band.** *Radio and Television.* a specific range of frequencies, esp. a set of radio frequencies, as HF, VHF, and UHF." Id. at 162 (emphasis in original). The underlying concept of "bandwidth" is that it represents a range of frequencies. The definitions of the terms "bandwidth" and "band" do not agree with Applicants' use of variance in dependent claims 14-17 and 20-23, which variance is used for determining whether a frequency-unit is an available frequency-unit.

bandwidth counter in step 2 of claim 5. There is only a recitation that a representative is obtained for each frequency-unit. There is no mention as to what the representative is used for or that anything else occurs with respect to the frequency-unit in step 2.[6]

Step 3 of claim 5 recites the "step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria." Once again, the Examiner has not directed Applicants' or the Board's attention as to where in the Veldhuis et al. patent or the Nishiguchi et al. patent there is a teaching of this step for counting the number of available frequency-units by using the representative.[7] As stated in the Appeal Brief at page 6, lines 17-21, the variance itself,

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[6]The Examiner is correct that the term "variance" is used in Applicants' dependent claims. *See* claims 14-17 and 20-23.

[7]Unless this is the step the Examiner is referring to with "assigning bandwidth" at the top of page 6 of the Examiner's Answer as quoted in the preceding paragraph of this Reply Brief. If this is correct, then the Examiner does not appreciate Applicants' use of an individual representative for each frequency-unit to determine if a particular frequency-unit is an available frequency-unit. "Bandwith" is defined as "*[t]elecommunications.* the smallest range of frequencies constituting a band, within which a particular signal can be transmitted without distortion." *Random House Unabridged Dictionary*, page 163 (2nd ed.) (emphasis in original). "Band" is "[a]lso called **frequency band, wave band.** *Radio and Television.* a specific range of frequencies, esp. a set of radio frequencies, as HF, VHF, and UHF." Id. at 162. (emphasis in original). The underlying concept of "bandwidth" is that it represents a range of frequencies. The frequency width of a particular frequency-unit is not recited as being impacted by the obtained representative for that particular frequency-unit. Furthermore, the location of the frequency-unit in terms of the frequency range covered is not recited as impacting whether the frequency-unit is an available frequency-unit.

contrary to what has been alleged by the Examiners, is not used to position frequency-units within a frequency interval. Consequently, a frequency-unit is not assigned to a particular frequency range based on the variance, but instead the variance as recited in the dependent claims is used for the representative in determining whether a frequency-unit is an available frequency-unit for the allocation of bits in the first bit allocation step.

The fourth step in claim 5 is the "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." (emphasis added). Key concepts of this step are the step is an "initial bit allocation" and the mathematical model uses the counted number of available frequency-units.

As is clear in claim 5, the first bit allocation step determines an initial bit allocation which, read in conjunction with the recitation of a second bit allocation step, provides two separate and distinct allocation steps. There is not one amorphous combination step as is present in the Nishiguchi et al. patent.[8] There is not a single correction step recited in claim 5, but two distinct allocation steps. Nishiguchi et al. patent teaches only one bit allocation step that is amorphously connected with a correction of the allocation. Applicants

[8]This argument about the Nishiguchi et al. patent disclosing only one step is developed more fully in discussing Applicants' second bit allocation step *infra*.

respectfully submit that this teaching of the Nishiguchi et al. patent does not make obvious the recitation of the two separate and distinct allocation steps, as recited in Applicants' claim 5.

The Examiner appears to have erroneously equated "mathematical model" with "mathematical algorithm" by stating that he "notes that a mathematical model or algorithm is a series of mathematical steps." Examiner's Answer, last line page 6 to page 7, line 1. As the Board is well aware, the term "mathematical algorithm" has become a term of art in patent law based on the various statutory subject matter cases matter under 35 U.S.C. §101, and consequently has a definition separate from that of the term "mathematical model". This Board has previously summarized the meaning of "mathematical algorithm" in *Ex parte Logan*, 20 USPQ2d 1465, 1467 (BPAI 1991), with the following:

> A mathematical algorithm is a "procedure for solving a given type of mathematical problem." *Gottschalk v. Benson*, 409 U.S. 63, 175 USPQ 673, 674 (1972); *Diamond v. Diehr*, 450 U.S. 175, 209 USPQ 1, 8 (1981). Mathematical algorithms include mathematical equations and formulas for calculating a numerical output value from a number of numerical input values, whether directly or indirectly claimed. A mathematical formula or equation is directly claimed when it is expressed in the claim by means of mathematical symbols, as in *Parker v. Flook*, 437 U.S. 584 (1978), and *Diehr, supra*. Indirect recitation can occur in either of two ways. The first, discussed in *re Freeman*, 573 F.2d 1237, 1246, 197 USPQ 464, 471 (CCPA 1978), is when a claim recites a mathematical formula or equation in terms of prose equivalents for mathematical symbols (e.g., the term "add" instead of the symbol "+"). The second is when the specification indicates a claimed step (e.g., "determining" or "cross-correlating") is implemented using a particular mathematical equation of formula. *Johnson, supra* at 1078, 200 USPQ at 208; *In re Sherwood*, 613 F.2d 809, 818, 204 USPQ 537, 545 (CCPA 1980).

The Board further simplified when a "mathematical algorithm" was being claimed, namely, "if it *essentially* recites, directly or indirectly, a method of computing one or more numbers from a different set of numbers by performing a series of mathematical computations." *Id.* at 1468 (emphasis in original).

Algorithm also has a dictionary definition which provides additional insight. "Algorithm" is defined as "a set of rules for solving a problem in a finite number of steps." *Random House Unabridged Dictionary*, p. 52 (2nd ed.).

In contrast, "mathematical model" has a different meaning as a term of art than "algorithm". "Model" has many different definitions.[9] The most relevant definition of "model" as a noun is "a simplified representation of a system or phenomenon, as in the sciences or economics, with any hypotheses required to describe the system or explain the phenomenon, often mathematically." *Id.* at 1235-6. The definition of "model" when it is a verb is "to simulate (a process, concept, or the operation of a system), commonly with the aid of a computer." *Id.* at 1236. One of ordinary skill in the art would understand the above selected definitions of "model" to adequately and accurately define "mathematical model" as the term is used within the scientific and engineering fields. A "mathematical model" is used to attempt to simulate and quantify with numbers a real

[9]There are 25 different definitions for the term "model" in the *Random House Unabridged Dictionary*, pages 1235-6 (2nd ed.).

world occurrence or system, while in contrast an "algorithm" provides a numerical solution or answer to a problem.

The Examiner is correct in noting that the Veldhuis et al. patent discloses an algorithm having a series of steps, as this term "algorithm" is known to one of ordinary skill in the art and as defined in the dictionary. Examiner's Answer, page 7, line 1. Applicants' claim 5, however, recites the use of a "mathematical model" which, as discussed above, is not the same as an "algorithm". The portion relied upon in the Veldhuis et al. patent by the Examiner teaches a comparison of a parameter to a threshold value to obtain a comparator signal. However, Applicants have noted in column 3 of the reference there is disclosed a means for determining an allocation pattern. It is this allocation pattern means that is an algorithm.

It is proper to attack the Veldhuis et al. patent on its alleged teaching of an algorithm as being unable to teach Applicants' recited mathematical model, because this feature is being taken from the Veldhuis et al. patent in combination with features from the Nishiguchi et al. patent to provide an overall combination that is alleged to teach or suggest Applicants' recited mathematical model. Furthermore, the Nishiguchi et al. patent does not teach or suggest Applicants' recited mathematical model using the counted number of available frequency-units and is unable to remedy this deficiency of the Veldhuis et al.

patent.[10] Therefore, when these two patents are combined, there is no teaching or suggestion of Applicants' method defined in claim 5 utilizing a mathematical model.

Applicants' last recited step in claim 5 is the "second bit allocation step for determining further bit allocation for each of said available frequency-units based on psychoacoustic criteria." The focus on this step has been on the words "second bit allocation step". Applicants submitted in the Appeal Brief at page 12, lines 8-18, that there is a difference between "allocating" and "correcting".

The Examiner appears to have interchanged the terms "allocate" and "correct" in rejecting Applicants' claims. Applicants respectfully submit that these two terms have two different and separate meanings. "Allocate", which is used in claim 5, is defined as "to set apart for a particular purpose; assign or allot: *to allocate funds for new projects.*" *Random House Unabridged Dictionary*, p. 56 (2nd ed.) (emphasis in original). In contrast, "correct" is defined as "to set or make true, accurate, or right; remove the errors or faults from" and "to counteract the operation or effect of (something hurtful or undesirable)." *Id.* at 454.

The underlying impact of these definitions is that bits could be allocated in one step and then corrected in yet another step similar to what is taught in the Nishiguchi et al. patent.

[10]This fact pattern is not the fact pattern of *In re Keller* as discussed *supra*.

This is not what is defined in Applicants' claim 5. Applicants' claim 5 recites two separate allocation steps where bits are allocated to frequency-units that are determined as being available in the third step of the claim. In complete contrast, the Nishiguchi et al. patent teaches that bits are allocated in a first step and then in the second step errors in the bit allocation are corrected to prevent something undesirable, such as too many bits being assigned to one frequency range.[11]

The Nishiguchi et al. patent states at col. 3, lines 10 that:

> a quantization circuit for quantizing the components of each band with the number of bits consistent with the level difference between the energy of each band and the output of the noise level setting means, wherein the volume of output information of that quantization circuit is detected and the bit allocation for the quantization circuit is corrected in dependence upon the error between the detection output and the target value to render the information volume constant over a predetermined time interval.

From the above excerpt it is clear that after an initial bit allocation there is a correction step based on an error, not a further bit allocation step as recited in Applicants' claim 5. As stated above, the term "correct" is not the same as the term "allocate", both in terms of definitions from the dictionary and the meaning one of ordinary skill in the art would understand these terms to have.

[11]As noted earlier, the Nishiguchi et al. patent actually teaches one step where the allocation is corrected prior to the bits actually being assigned.

Furthermore, one could argue that the Nishiguchi et al. patent teaches only one step that corrects the anticipated bit allocation prior to actually allocating bits to each band. The Nishiguchi et al. patent, in fact, suggests as much with the following:

> if the level *a* corresponding to the allowable noise level, as later described, for computing the masking spectrum of the Burke spectrum SB, that is the allowable noise spectrum, is low, the masking level or the masking curve is decreased, so that it becomes necessary to increase the number of bits allocated at the time of quantization of the quantization circuit 24. Conversely, if the level *a* is high, the masking level is increased, with the result that the number of bits allocated at the time of quantization may be diminished.

Nishiguchi et al., col. 6, lines 30-39. In which case, one step is unable to teach Applicants' two steps. The above excerpt also supports Applicants' position that the Nishiguchi et al. device corrects a bit allocation, but does not use two separate bit allocation steps.

As stated previously, it is appropriate to attack an individual reference when that reference is relied upon to teach a particular feature. The Examiner has relied upon the Nishiguchi et al. patent to teach or suggest two "bit allocation" steps. Contrary to the Examiner's position, the Nishiguchi et al. patent teaches at most only one bit allocation step and one correction step which are different from Applicants' two bit allocation steps. The Examiners handling this application have admitted that the Veldhuis et al. patent does not teach two bit

allocation steps,[12] and the Nishiguchi et al. patent is therefore unable to remedy this deficiency, for the reasons stated above.[13]

Applicants respectfully traverse the Examiner's reasons that the combination of the Veldhuis et al. and Nishiguchi et al. patents is a valid combination for the reasons discussed in Applicants' Appeal Brief from page 10, line 24 to page 11, line 16 and from page 12, line 22 to page 13, line 13. Applicants also submit the language of claim 5 inherently allows for any frequency-unit to be an available frequency-unit depending on the representative and psychoacoustic criteria used, contrary to the statement by the Examiner.[14]

Applicants respectfully submit that even if this combination of the Veldhuis et al. and Nishiguchi et al. patents is determined to be proper, this combination is still unable to teach or suggest each and every feature of Applicants' claim 5. Based on the plain meaning of the words used in claim 5, there is no way that the recited features are taught or suggested by the Veldhuis et al. patent in combination with the Nishiguchi et al. patent. In particular, the Examiner has failed to create a

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[12] *See* January 6, 1997 Office Action, pages 5-6, paragraph 7 (where the Examiner modified the earlier 35 U.S.C. §102 rejection into a 35 U.S.C. §103 rejection by combining Veldhuis et al. and Nishiguchi et al.); December 9, 1997 Examiner's Answer, pages 6 and 8.

[13] Once again, this fact pattern is not the fact pattern of *In re Keller* as previously discussed.

[14] Examiner's Answer page 8, lines 13-17.

combination that teaches the counting of a number of available frequency-units based on psychoacoustic criteria in conjunction with the obtained representative, a first initial bit allocation step that uses an approximate mathematical model that relies upon the counted number of available frequency-units, and a second bit allocation step. Therefore, Applicants respectfully submit that these claims are patentable over the combination of the Veldhuis et al. patent and the Nishiguchi et al. patent.

For the above reasons and the reasons stated in Applicants' Appeal Brief, Applicants' claimed invention is not taught or suggested by the Veldhuis et al. and Nishiguchi et al. patents. All of the specific limitations in the claims are not described in the prior art relied upon in this rejection. Furthermore, the combination of these two patents does not teach or suggest the recited invention. Therefore, the Examiner has made an improper rejection of Applicants' recited invention based on a combination of the Veldhuis et al. and Nishiguchi et al. patents. Therefore, the rejected claims 4-7, 9-12, 14-17, and 19-23 defining Applicants' invention are patentable.

Reversal of the final rejection is earnestly solicited.

Respectfully submitted,

BEVERIDGE, DEGRANDI, WEILACHER & YOUNG, L.L.P.

by: _____

Joseph A. DeGrandi, Reg. No. 17,446
1850 M Street, N.W. - Suite 800
Washington, D.C. 20036
Telephone: (202) 659-2811
Facsimile: (202) 659-1462

February 9, 1998

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PATENT

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE
BEFORE THE BOARD OF PATENT APPEALS AND INTERFERENCES

Applicant : Do Hui TEH et al. Examiner: M. Opsasnick

Serial No. : 08/161,798 Group Art Unit: 2741

Filed : December 6, 1993

For : A BIT ALLOCATION METHOD FOR DIGITAL AUDIO SIGNALS (AS AMENDED)

REQUEST FOR ORAL HEARING

Assistant Commissioner for Patents
Washington, D.C. 20231

Sir:

In accordance with 37 C.F.R. §1.194, Applicant respectfully requests an oral hearing for the pending appeal in the above-identified patent application.

Enclosed herewith is a check in the amount of $270.00 to cover the fee for this Request in accordance with 37 C.F.R. §1.194(b) and 37 C.F.R. §1.17(d). Please charge any additional fees which may be necessary to Deposit Account No. 02-4300, and credit any overpayment to said Deposit Account.

Respectfully submitted,

BEVERIDGE, DeGRANDI, WEILACHER & YOUNG, L.L.P.

By: Joseph A. DeGrandi

Joseph A. DeGrandi, Reg. No. 17,446
1850 M Street, N.W., Suite 800
Washington, D.C. 20036
Telephone: (202) 659-2811
Facsimile: (202) 659-1462

February 9, 1998

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BEVERIDGE, DEGRANDI, WEILACHER & YOUNG
1850 M STREET, N. W., SUITE 800
WASHINGTON DC 20036

EXAMINER	
OPSASNICK, M	
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Application No. 08/161798

Reply to Reply Brief

1) The reply brief filed 2/9/98 has been entered and considered. The application has been forwarded to the Board of Patent Appeals and Interferences for decision on the appeal.

2. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Michael Opsasnick whose telephone number is (703)305-4089.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, David R. Hudspeth, can be reached at (703)308-4825. The facsimile phone number for this group is (703)305-9508.

Any inquiry of a general nature or relating to the status of this applications should be directed to the Group receptionist whose telephone number is (703)305-3900.

DAVID R. HUDSPETH
SUPERVISORY PATENT EXAMINER
GROUP 2700

Michael N. Opsasnick

April 21, 1998

OIPE JC89
OCT 28 1999
PATENT & TRADEMARK OFFICE

Attorney Docket No.
03-48683

PATENT

RECEIVED
OCT 29 1999
TECH CENTER 2700

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicant : TEH ET AL.

Serial No. : 08/161,798 Examiner: M. OPSASNICK

Filed : December 6, 1993 Group Art Unit: 2308

For : BIT ALLOCATION METHOD FOR TRANSFORM CODER

COMBINED PETITION UNDER 37 C.F.R. SECTION 1.97(d)(2)
AND STATEMENT UNDER 37 C.F.R. SECTION 1.97(e)(1)

This paper respectfully requests consideration of an Information Disclosure Statement (IDS) that is submitted concurrently herewith. This paper constitutes a petition under 37 C.F.R. Section 1.97(d)(2) for requesting such consideration. The Petition Fee of $130 is submitted herewith by the undersigned firm's check in that amount.

The information contained in the accompanying IDS derives from a communication from the Japanese Patent Office (JPO) issued for the Applicants' counterpart application before the JPO. The accompanying IDS explains that it is submitting an EP laid-open application as the English language counterpart for the JP laid-open application that was cited in the office action from the JPO. The JPO office action is dated August 10, 1999. As such, this combined Statement and Petition, and the accompanying IDS are each filed before three months after August 10, 1999.

1999 NOV -3 AM 8:55
BOARD OF PATENT APPEALS AND INTERFERENCES

In view of the foregoing, entry and consideration of the accompanying IDS are solicited.

Respectfully submitted,

SMITH, GAMBRELL & RUSSELL, LLP

Beveridge, DeGrandi, Weilacher & Young
Intellectual Property Group

By: _____

Michael A. Makuch, Reg. No. 32,263
1850 M Street, N.W., Suite 800
Washington, D.C. 20036
Telephone: (202) 659-2811
Facsimile: (202) 659-1462

October 28, 1999

BOARD OF PATENT APPEALS AND INTERFERENCES
1999 NOV -3 AM 8:55

GAP B68
8741 #19



Attorney Docket No.
03-48683

PATENT

H.D.S.
11-9-99
hmb

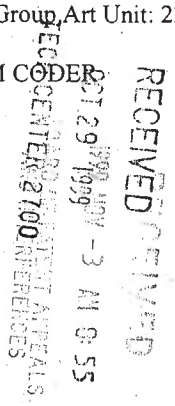
IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicant : TEH ET AL.

Serial No. : 08/161,798 Examiner: M. OPSASNICK

Filed : December 6, 1993 Group Art Unit: 2308

For : BIT ALLOCATION METHOD FOR TRANSFORM CODER



INFORMATION DISCLOSURE STATEMENT

Attn.: Box Applications

Assistant Commissioner for Patents
and Trademarks
Washington, D.C. 20231

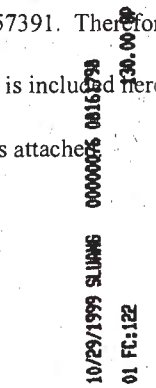
Sir:

Pursuant to their duty of disclosure under 37 C.F.R. § 1.56, Applicants bring the following document to the attention of the Examiner in the above-identified patent application:

1. European Patent Application No. 0 457391.

Applicants just recently received an official action from the Japanese Patent Office (JPO) for their Japanese counterpart to the above-identified application. A copy of the JPO office action is attached. The action is dated August 10, 1999.

The JPO action cited JP laid-open 4-250723. However, there is an English language counterpart to JP laid-open 4-250723. This is EP (laid-open) 0 457391. Therefore, the English language EP application is submitted herewith. A PTO-1449 form is included herewith listing the EP laid-open document, and a copy of this laid-open document is attached.



OIPE JC599
OCT 28 1999
PATENT & TRADEMARK OFFICE

Because this Statement is being filed in connection with an application under appeal of a final rejection, a Combined Petition and Statement in accordance with 37 C.F.R. Section 1.97(d)(2) and Section 1.97(e)(1) is being submitted along with the requisite fee for such Petition.

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OCT 29 1999
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Respectfully submitted,

SMITH, GAMBRELL & RUSSELL, LLP

Beveridge, DeGrandi, Weilacher & Young
Intellectual Property Group

By: _____

Michael A. Makuch, Reg. No. 32,263
1850 M Street, N.W., Suite 800
Washington, D.C. 20036
Telephone: (202) 659-2811
Facsimile: (202) 659-1462

October 28, 1999

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BOARD OF PATENT APPEALS
AND INTERFERENCES

FORM PTO-1449

INFORMATION DISCLOSURE STATEMENT

(Use several sheets if necessary)

OCT 2 8 1998 PATENT & TRADEMARK OFFICE JC89

ATTY. DOCKET NO.	SERIAL NO.
03-48683	08/161,798
APPLICANT: TEH ET AL.	
FILING DATE December 6, 1993	GROUP ART UNIT ~~2308~~ 2748

*Examiner's Initials		DOCUMENT NUMBER	DATE	NAME	CLASS	SUB-CLASS	FILING DATE, IF APPROPRIATE
	AA						
	AB						
	AC						
	AD						
	AE						
	AF						
	AG						

RECEIVED OCT 29 1998 TECH CENTER 2700

*Examiner's Initials		DOCUMENT NUMBER	DATE	COUNTRY	CLASS	SUB-CLASS	TRANSLATION YES	TRANSLATION NO
MO	AH	0457391	November 21, 1991	Europe	—	—	x	
	AI							
	AJ							

OTHER INFORMATION (Including Author, Title, Date, Pertinent Pages, Etc.)		
	AK	
	AL	
	AM	
	AN	
	AO	
	AP	
	AQ	
	AR	
	AS	
	AT	
	AU	
	AV	
	AW	

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EXAMINER M. M. Gravini	DATE CONSIDERED 1/24/00

*EXAMINER: Initial if reference considered, whether or not citation is in conformance with MPEP 609; Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

Paper No. 20

UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE BOARD OF PATENT APPEALS
AND INTERFERENCES

Ex parte DO HUI TEH, PEK YEW TAN and SUA HONG NEO

Appeal No. 1999-1793
Application No. 08/161,798[1]

MAILED

NOV 2 4 1999

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BOARD OF PATENT APPEALS
AND INTERFERENCES

ORDER REMANDING TO EXAMINER

An Information Disclosure Statement (IDS) was filed October 28, 1999 (Paper No. 19) and has been matched with this application at the Board of Patent Appeals and Interferences. The IDS needs to be considered by the Primary Examiner with respect to compliance with the criteria set forth in 37 C.F.R. §§ 1.97 and 1.98. A communication notifying applicant of the Primary Examiner's decision is required.

[1] Application for patent filed December 6, 1993.

Accordingly, it is

ORDERED that the application is remanded to the Examiner for such consideration of the IDS, appropriate notification to applicant and for such further action as may be appropriate.

BOARD OF PATENT APPEALS
AND INTERFERENCES

By: _____
CRAIG R. FEINBERG
Program and Resource Administrator
(703) 308-9797

bae

cc: Beveridge, DeGrandi, Weilacher & Young
1850 M Street, NW
Suite 800
Washington, DC 20036

Transaction History Date 2000-02-01
Date information retrieved from USPTO Patent Application Information Retrieval (PAIR) system records at www.uspto.gov

UNITED STATES DEPARTMENT OF COMMERCE
Patent and Trademark Office
Address: COMMISSIONER OF PATENTS AND TRADEMARKS
Washington, D.C. 20231

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.
08/161,795	12/06/93	TEH	

LM41/0201

REVERIDGE, DEGRANDI, WEILACHER & YOUNG
1850 M STREET, N. W., SUITE 800
WASHINGTON DC 20036

EXAMINER	
OPSASNICK, M	
ART UNIT	**PAPER NUMBER**
2708	

DATE MAILED: 02/01/94

Please find below and/or attached an Office communication concerning this application or proceeding.

Commissioner of Patents and Trademarks

PTO-90C (Rev. 2/95)
*U.S. GPO: 1997-422-198/60031

1- File Copy

DETAILED ACTION

Information Disclosure Statement

1. The information disclosure statement submitted on October 28, 1999 was filed after the mailing date of the Final Office Action on January 6, 1997. The submission is in compliance with the provisions of 37 CFR 1.97 and 37 CFR 1.98. Accordingly, the petition is granted and the information disclosure statement ~~is being~~ has been considered by the examiner.

KLT 4-27-00

2. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Michael Opsasnick whose telephone number is (703)305-4089.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Ms. Krista Zele, can be reached at (703)305-4701. The facsimile phone number for this group is (703)308-6296.

Any inquiry of a general nature or relating to the status of this applications should be directed to the Group receptionist whose telephone number is (703)305-3900.

Michael N. Opsasnick

January 24, 2000

KRISTA ZELE
SUPERVISORY PATENT EXAMINER
GROUP 2700

Transaction History Date 2000-05-01
Date information retrieved from USPTO Patent Application Information Retrieval (PAIR) system records at www.uspto.gov

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UNITED STATES DEPARTMENT OF COMMERCE
Patent and Trademark Office
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Washington, D.C. 20231

Paper No. 22(TCS)

Beveridge, DeGrandi, Weilacher & Young
1850 M Street, NW, Suite 800
Washington, DC 20036

Appeal No.: 99-1793
Appellant: Teh, Do Hui, et.al
Serial No.: 08/161,798

Hearing: Room B
Docket: B
Date: June 8, 2000
Time: 9:00 a.m.
Place: Room 12C07
CRYSTAL GATEWAY 2
1225 Jefferson Davis Highway
Arlington, VA 22202

NOTICE OF HEARING

CONFIRMATION REQUIRED WITHIN TWENTY-ONE DAYS

Your attention is directed to 37 CFR § 1.194(a).

The above identified appeal will be heard by the Board of Patent Appeals and Interferences on the date indicated. Hearings will commence at the time set and as soon as the argument in one appeal is concluded, the succeeding appeal will be taken up

The time allowed for argument is twenty minutes unless additional time is requested and permitted before the argument is commenced.

CONFIRMATION OR WAIVER OF THE ORAL HEARING IS REQUIRED. This form must **BE COMPLETED BELOW** and filed with the Board of Patent Appeals and Interferences preferably by facsimile within **TWENTY-ONE (21) DAYS** from the mailing date of this notice, indicating confirmation or waiver of the hearing. A copy of this form may alternatively be filed by mail if facsimile is not available

Failure to file this form within the above time period will be construed as a waiver of the request for oral hearing

37 CFR § 1.136(a) does not apply.

By Order of the Board of Patent Appeals and Interferences

B.P.A.I. FAX No. is:
(703) 308-7952 or (703) 308-7953
See 1108 Off. Gaz. Pat. & Trademark Office 15 (Nov.14, 1989)

B.P.A.I. Mailing Address is:
BOARD OF PATENT APPEALS AND INTERFERENCES
COMMISSIONER OF PATENT AND TRADEMARKS
WASHINGTON, D.C. 20231

Clerk of the Board
(703) 308-9797

In all communications relating to this appeal, please identify the appeal by its number.

CHECK ONE : ☐ **HEARING ATTENDANCE CONFIRMED.**

☐ **HEARING WAIVED.**

_____ _____ _____
Signature of Attorney/Agent/Appellant | Date | Registration No.

FORM PTOL-38
REV. 1/99)

5.22.00 CONFIRMATION DUE BY

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TCS

Paper No. 22(TCS)

Beveridge, DeGrandi, Weilacher & Young
1850 M Street, NW, Suite 800
Washington, DC 20036

Appeal No.: 99-1793
Appellant: Teh, Do Hui, et al
Serial No.: 08/161,798

Hearing: Room B
Docket: B
Date: June 8, 2000
Time: 9:00 a.m.
Place: Room 12C07
CRYSTAL GATEWAY 2
1225 Jefferson Davis Highway
Arlington, VA 22202

NOTICE OF HEARING

CONFIRMATION REQUIRED WITHIN TWENTY-ONE DAYS

Your attention is directed to 37 CFR § 1.194(a).

The above identified appeal will be heard by the Board of Patent Appeals and Interferences on the date indicated. Hearings will commence at the time set and as soon as the argument in one appeal is concluded, the succeeding appeal will be taken up

The time allowed for argument is twenty minutes unless additional time is requested and permitted before the argument is commenced.

CONFIRMATION OR WAIVER OF THE ORAL HEARING IS REQUIRED. This form must **BE COMPLETED BELOW** and filed with the Board of Patent Appeals and Interferences preferably by facsimile within **TWENTY-ONE (21) DAYS** from the mailing date of this notice, indicating confirmation or waiver of the hearing. A copy of this form may alternatively be filed by mail if facsimile is not available

Failure to file this form within the above time period will be construed as a waiver of the request for oral hearing

37 CFR § 1.136(a) does not apply.

By Order of the Board of Patent Appeals and Interferences

B.P.A.I. FAX No. is:
(703) 308-7952 or (703) 308-7953
See 1108 Off. Gaz. Pat. &
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B.P.A.I. Mailing Address is:
BOARD OF PATENT APPEALS AND INTERFERENCES
COMMISSIONER OF PATENT AND TRADEMARKS
WASHINGTON, D.C. 20231

Clerk of the Board
(703) 308-9797

In all communications relating to this appeal, please identify the appeal by its number.

CHECK ONE : ☐ **HEARING ATTENDANCE CONFIRMED.**

☒ **HEARING WAIVED.**

Signature of Attorney/Agent/Appellant
Michael A. Makuch

22 May 2000
Date

32,263
Registration No.

The opinion in support of the decision being entered today was not written for publication and is not binding precedent of the Board.

Paper No. 24

UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE BOARD OF PATENT APPEALS
AND INTERFERENCES

Ex parte DO HUI TEH,
PEK YEW TAN,
and SUA HONG NEO

MAILED

JUL 30 2001

PAT. & T.M. OFFICE
BOARD OF PATENT APPEALS
AND INTERFERENCES

Appeal No. 1999-1793
Application 08/161,798[1]

ON BRIEF

Before HAIRSTON, JERRY SMITH, and BARRETT, Administrative Patent Judges.

BARRETT, Administrative Patent Judge.

DECISION ON APPEAL

This is a decision on appeal under 35 U.S.C. § 134 from the final rejection of claims 4-7, 9-12, 14-17, and 19-23. Claims 8, 13, 18, and 24 are allowed. Claims 1-3 are canceled.

We reverse.

[1] Application for patent filed December 6, 1993, entitled "Bit Allocation Method For Digital Audio Signals," which claims the foreign filing priority benefit under 35 U.S.C. § 119 of Japanese Application 5-030712, filed February 19, 1993.

BACKGROUND

The disclosed invention relates to a dynamic bit allocation method that is adapted to the human auditory system and yet has a low level of complexity. The complexity is reduced by allocating an initial bit number using approximations made in a mathematical model before the remaining bit numbers are allocated by a precise iterative process.

Claim 5 is reproduced below.

> 5. 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:
>
> a step for dividing at least one frequency interval into a plurality of frequency-units;
>
> a step for obtaining a representative of each frequency-unit obtained;
>
> a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria;
>
> 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
>
> a second bit allocation step for determining further bit allocation for each of said available frequency-units based on psychoacoustic criteria.

The Examiner relies on the following prior art:

Veldhuis et al. (Veldhuis)	5,105,463	April 14, 1992
Nishiguchi et al. (Nishiguchi)	5,151,941	September 29, 1992

Claims 4[2]-7, 9-12, 14-17, and 19-23 stand rejected under 35 U.S.C. § 103(a) as being unpatentable over Veldhuis and Nishiguchi.

We refer to the rejection (Paper No. 5), the final rejection (Paper No. 8) (pages referred to as "FR__") and the examiner's answer (Paper No. 15) (pages referred to as "EA__") for a statement of the Examiner's position, and to the appeal brief (Paper No. 14) (pages referred to as "Br__") and the reply brief (Paper No. 16) (pages referred to as "RBr__") for Appellants' arguments thereagainst.

OPINION

The claims are grouped to stand or fall together with independent claim 5 (Br5).

Appellants argue that the Examiner has improperly read limitations into claim 5 (e.g., Br7-8; RBr5-6). To the extent this is so, we address only the actual limitations of claim 5.

The Examiner's[3] discussion of Veldhuis (FR5) is very cursory and does not correlate the actual limitations of claim 5 to the disclosure of Veldhuis. Therefore, we begin by making findings regarding the contents of Veldhuis and the differences between Veldhuis and the subject matter of claim 5.

---

[2] The Examiner's statement of the rejection inadvertently omits claim 4.

[3] As noted by Appellants (RBr2 n.1), there have been numerous examiners during prosecution.

Claim 5 recites "buffering audio signals in frames and decomposing the digital signals into spectral components" and "a step for dividing at least one frequency interval into a plurality of frequency-units." A "unit" is a frequency interval (specification, p. 8). Veldhuis does not disclose that the audio signal is buffered in frames; however, while this limitation is considered either inherent or obvious to one of ordinary skill in the art, it is not argued and will not be addressed. See 37 CFR § 1.192(c)(8)(iv) (1997) (brief must specify the errors in the rejection). Veldhuis discloses that the audio signal x(k) is applied to an analysis filter bank 3 which divides the signal band of 0-22.05 kHz into P=26 subbands (col. 5, lines 58-64), the characteristics of which are shown in the table of figure 3. The bandwidths W(p) of the subbands approximately correspond to the critical bandwidths of the human auditory system in the respective frequency ranges (col. 6, lines 40-46). Thus, we find Veldhuis discloses "decomposing the digital signals into spectral components" and "a step for dividing at least one frequency interval into a plurality of frequency-units." 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. Thus, Appellants' arguments that the lower subbands do not use adaptive bit allocation and that their invention

allows for the possibility of assigning and allocating bits to the whole frequency spectrum (e.g., Br10-13) are not persuasive.

Claim 5 recites "a step for obtaining a representative of each frequency-unit obtained." The term "representative" is not expressly defined in the specification. However, from the statement "the variance or a representative within a defined frequency interval as an accurate representation of the signals in the interval" (specification, p. 5), we interpret "representative" to mean a "characteristic representative of the signal." Veldhuis discloses a signal buffer 6(p) for each subband and a level detector 7(p) connected to each signal buffer 6(p) to determine for each block stored having block number m a characteristic parameter G(p;m) representative of the signal level of the block (col. 6, lines 17-23). The signal level can be represented by the average value of the amplitude or the power of the signal samples of a block, and also by the peak value of the amplitude of the signal samples in a block (col. 6, lines 29-33). The set of "characteristic parameters G(p;m)" in Veldhuis is the "representative of each frequency-unit obtained." Thus, Veldhuis discloses "a step for obtaining a representative of each frequency-unit obtained."

Claim 5 recites "a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria." For the upper group of subbands

($13 \leq p \leq 26$) in Veldhuis, the number of quantizing bits per signal sample $B(p;m)$ is determined by the set of all characteristic parameters $G(p;m)$ (col. 10, lines 8-18). As previously noted, the "at least one frequency interval" in the first step of claim 5 reads on the frequency interval corresponding to the upper subbands ($13 \leq p \leq 26$) in Veldhuis, so it is proper to focus on these subbands (frequency-units). Parameter $G(p;m)$ is compared to the threshold $T(p)$ for the subband of band number p and a binary comparator signal $C(p;m)$ is generated having a first value $C(p;m)$="1" for $G(p;m) \geq T(p)$ and a second value $C(p;m)$="0" for $G(p;m) < T(p)$ (col. 11, lines 31-36). The thresholds are related to the thresholds of the human auditory system for perceiving single tones in the respective frequency ranges (col. 11, lines 36-44); thus, the thresholds are "based on psychoacoustic criteria." By these comparisons, the blocks (p;m) of subband signals $x_p(k)$ are divided into blocks (p;m) containing perceptually significant signal energy on the basis of the criterion $G(p;m) \geq T(p)$ and thus having the value $C(p;m)$="1", and blocks (p;m) containing no perceptually significant energy on the basis of the criterion $G(p;m) < T(p)$ and thus having a value of $C(p;m)$="0" (col. 11, lines 45-52). No quantizing bits are allocated to the blocks (p;m) within the same allocation window having $C(p;m)$="0" and the quantizing bits thus saved are used for the finer quantization of the blocks (p;m) within the same

allocation window having C(p;m)="1" (col. 11, lines 52-57). This eliminates blocks (p;m) that are irrelevant. The blocks (p;m) having the value C(p;m)="1" are "available frequency-units" because they are available for bit allocation. The comparator signal C(p;m), which determines whether a block (p;m) is "available", is based on the threshold T(p) for the subband, which is "based on psychoacoustic criteria"; thus, the determination of available frequency-units is based on psychoacoustic criteria. The number of blocks (p;m) in the allocation window having C(p;m)="1" is the count of available frequency-units. However, Veldhuis does not disclose "counting" the available blocks (p;n). Therefore, Veldhuis does not disclose "a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria" (emphasis added).

Claim 5 recites "a first bit allocation step for determining an initial bit allocation for each of said available frequency-units by using an approximate mathematical model using said counted number of available frequency-units." The blocks (p;m) in the allocation window having C(p;m)="1" are the "available frequency-units." All of the blocks are initially allocated a minimum number of bits B(p;m) as a default value for the allocation pattern (e.g., col. 13, lines 60-64; col. 21, lines 35-37) and then those blocks (p;m) with C(p;m)="0" are set to B(p;m)=0 to produce sum S of "saved" number of bits (col. 13,

line 68 to col. 14, line 12), leaving the initial allocations in blocks (p;m) with C(p;m)="1". This is a first bit allocation step for each available frequency-unit. However, this initial bit allocation is not disclosed or suggested to be made "using an approximate mathematical model using said counted number of available frequency-units." Therefore, we find Veldhuis 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" (emphasis added)

Claim 5 lastly recites "a second bit allocation step for determining further bit allocation for each of said available frequency-units based on psychoacoustic criteria." As disclosed, the second step of allocating bits "based on psychoacoustic criteria" uses psychoacoustic criteria only in the sense that a psychoacoustic weighting factor F is used in reducing the variance of the unit (specification, p. 9, step [4]); thus, the allocation of bits "based on psychoacoustic criteria" can be interpreted broadly as very indirectly based on some psychoacoustic criterion. In Veldhuis, the blocks (p;m) having the greatest need of quantizing bits G'(p;m) are allocated further bits by jumping to the next level i in figure 10 (step 9-11 in fig. 9; col. 15, lines 58-63). The need of bits G'(p;m) is assigned a value of G'(p;m)/D, with D>1, so as to evenly

distribute the numbers of quantization bits saved over the blocks (p;m) having a value of C(p;m)="1" (col. 15, line 63 to col. 16, line 1). The value D is not disclosed to be "based on psychoacoustic criteria." Therefore, we find that Veldhuis does not teach "a second bit allocation step for determining further bit allocation for each of said available frequency-units based on psychoacoustic criteria" (emphasis added).

Based on the above analysis, we summarize that the differences between the subject matter of claim 5 and Veldhuis are that Veldhuis does not teach: (1) "a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria" (emphasis added); (2) "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" (emphasis added); and (3) "a second bit allocation step for determining further bit allocation for each of said available frequency-units based on psychoacoustic criteria" (emphasis added).

The Examiner has further applied Nishiguchi. The Examiner finds that "Nishiguchi et al. teach a method for determining a bit allocation comprising a first and second bit allocation which incorporates psychoacoustic criteria (their initial allocation followed by a 'correcting step' col. 2 line 64- col. 3 line 14;

and psychoacoustics col. 4)" (FR6). The Examiner concludes that it would have been obvious to use the two-step process of Nishiguchi in the adaptive bit allocation taught by Veldhuis "because the number of available bits can change dynamically which results in bits being in surplus or deficit and leading to less efficient bit allocation, thus, requiring more bandwidth (Nishiguchi et al., col. 1)" (FR6).

We analyze the three differences noted above.

Difference (1) - counting available frequency-units

Appellants argue that the Examiner has not shown where the step of counting the number of available frequency-units is found in Veldhuis or Nishiguchi (RBr6). We do not find any discussion of the counting step by the Examiner.

We agree that neither Veldhuis nor Nishiguchi discloses counting the number of available frequency-units. In Veldhuis, the blocks (p;m) in the allocation window having C(p;m)="1" are "available frequency-units." However, Veldhuis does not disclose "counting" the number of available blocks (p;n) having C(p;m)="1". We do not find any disclosure of counting available frequency-units in Nishiguchi. Thus, the counting step is a difference which is not accounted for in the rejection.

Difference (2) - mathematical model using counted number of available frequency-units

The Examiner finds that Veldhuis teaches a "mathematical model" and points to Veldhuis at column 2, lines 54-64, showing B(p), G(p;m), and C(p;m) (EA6). The Examiner states that a mathematical model or algorithm is a series of mathematical steps as shown in Veldhuis at figures 8 and 9 (EA7).

Appellants respond that a "mathematical model" is not the same thing as a "mathematical algorithm" (RBr8-10) and that Nishiguchi does not teach or suggest a mathematical model using the counted number of available frequency-units (RBr10).

We agree with Appellants that a "mathematical model" is not necessarily a "mathematical algorithm." However, it is not necessary to explore the nuances in any detail since the Examiner has not shown a mathematical algorithm "using said counted number of available frequency-units," as claimed. The initial bit allocations in Veldhuis are all predetermined. Thus, the mathematical model using the counted number of frequency-units is a difference which is not accounted for in the rejection.

Difference (3) - second step of bit allocation based on psychoacoustic criteria

Appellants argue that the second step of further bit allocation in claim 5 is different than the second step of bit correction in Nishiguchi (Br12; RBr11-12). Appellants argue that Nishiguchi teaches only one bit allocation step that is

amorphously connected with a correction of the allocation and that Nishiguchi does not make obvious two separate and distinct allocation steps (RBr7-8).

The Examiner disagrees (EA7), but we do not understand the Examiner's rationale.

The arguments are directed to the existence of a two step allocation process, an initial allocation followed by a further bit allocation, rather than to the limitation of the further bit allocation being based on psychoacoustic criteria. We have found that Veldhuis discloses a two step process. Since the limitation about further allocation "based on psychoacoustic criteria" is not argued, it is not addressed. See 37 CFR § 1.192(c)(8)(iv).

CONCLUSION

Because the combination of Veldhuis and Nishiguchi does not disclose at least the limitations of "a step for counting a number of available frequency-units by using the representative, based on psychoacoustic criteria" (emphasis added) and "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" (emphasis added), the Examiner has failed to establish a prima facie case of obviousness. The rejection of claims 4-7, 9-12, 14-17, and 19-23 is reversed.

REVERSED

KENNETH W. HAIRSTON
Administrative Patent Judge

JERRY SMITH
Administrative Patent Judge

LEE E. BARRETT
Administrative Patent Judge

BOARD OF PATENT
APPEALS
AND
INTERFERENCES

Appeal No. 1999-1793
Application 08/161,798

BEVERIDGE, DeGRANDI, WEILACHER & YOUNG, L.L.P.
1850 M Street, N.W.
Suite 800
Washington, DC 20036

Transaction History Date 2001-09-28
Date information retrieved from USPTO Patent Application Information Retrieval (PAIR) system records at www.uspto.gov

Notice of Allowability	Application No. 08/161,798	Applicant(s) Teh et al
	Examiner Michael N. Opsasnick	Art Unit 2645

--The MAILING DATE of this communication appears on the cover sheet with the correspondence address

All claims being allowable, PROSECUTION ON THE MERITS IS (OR REMAINS) CLOSED in this application. If not included herewith (or previously mailed), a Notice of Allowance and Issue Fee Due or other appropriate communication will be mailed in due course. THIS NOTICE OF ALLOWABILITY IS NOT A GRANT OF PATENT RIGHTS. This application is subject to withdrawal from issue at the initiative of the Office or upon petition by the applicant. See 37 CFR 1.313 and MPEP 1308.

1. ☒ This communication is responsive to *Board Decision 7/30/01*
2. ☒ The allowed claim(s) is/are *5-24*
3. ☐ The drawings filed on _____ are acceptable as formal drawings.
4. ☒ Acknowledgement is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d).
   a) ☒ All b) ☐ Some* c) ☐ None of the:
   1. ☒ Certified copies of the priority documents have been received.
   2. ☐ Certified copies of the priority documents have been received in Application No. _____
   3. ☐ Copies of the certified copies of the priority documents have been received in this national stage application from the International Bureau (PCT Rule 17.2(a)).

   *Certified copies not received: _____
5. ☐ Acknowledgement is made of a claim for domestic priority under 35 U.S.C. § 119(e).

Applicant has THREE MONTHS FROM THE "MAILING DATE" of this communication to file a reply complying with the requirements noted below. Failure to timely comply will result in ABANDONMENT of this application. **THIS THREE-MONTH PERIOD IS NOT EXTENDABLE FOR SUBMITTING NEW FORMAL DRAWINGS, OR A SUBSTITUTE OATH OR DECLARATION. This three-month period for complying with the REQUIREMENT FOR THE DEPOSIT OF BIOLOGICAL MATERIAL is extendable under 37 CFR 1.136(a).**

6. ☐ Note the attached EXAMINER'S AMENDMENT or NOTICE OF INFORMAL APPLICATION (PTO-152) which gives reason(s) why the oath or declaration is deficient. A SUBSTITUTE OATH OR DECLARATION IS REQUIRED.
7. ☐ Applicant MUST submit NEW FORMAL DRAWINGS
   (a) ☐ including changes required by the Notice of Draftsperson's Patent Drawing Review (PTO-948) attached
      1) ☐ hereto or 2) ☐ to Paper No. _____.
   (b) ☐ including changes required by the proposed drawing correction filed _____, which has been approved by the examiner.
   (c) ☐ including changes required by the attached Examiner's Amendment/Comment or in the Office action of Paper No. _____.

   Identifying indicia such as the application number (see 37 CFR 1.84(c)) should be written on the drawings. The drawings should be filed as a separate paper with a transmittal letter addressed to the Official Draftsperson.
8. ☐ Note the attached Examiner's comment regarding REQUIREMENT FOR THE DEPOSIT OF BIOLOGICAL MATERIAL.

Any reply to this letter should include, in the upper right hand corner, the APPLICATION NUMBER (SERIES CODE/SERIAL NUMBER). If applicant has received a Notice of Allowance and Issue Fee Due, the ISSUE BATCH NUMBER and DATE of the NOTICE OF ALLOWANCE should also be included.

Attachment(s)

1 ☐ Notice of References Cited (PTO-892)
2 ☐ Notice of Informal Patent Application (PTO-152)
3 ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
4 ☐ Interview Summary (PTO-413), Paper No. _____
5 ☐ Information Disclosure Statement(s) (PTO-1449), Paper No(s). _____
6 ☐ Examiner's Amendment/Comment
7 ☐ Examiner's Comment Regarding Requirement for Deposit of Biological Material
8 ☒ Examiner's Statement of Reasons for Allowance
9 ☐ Other

FAN TSANG
SUPERVISORY PATENT EXAMINER
TECHNOLOGY CENTER 2600

DETAILED ACTION

Allowable Subject Matter

1. Claims 5-24 are allowable over the prior art of record.

2. The following is an examiner's statement of reasons for allowance:

As per the Board of Patent Appeals and Interferences, independent claims 5 and 8 recite the specific limitations pertaining to a step for counting a number of available frequency units by using the representative, based on psychoacoustic criteria, and 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, is not explicitly taught by the prior art of record (in particular, Veldhuis, and Nishiguchi). See Paper #24, pgs 1-13 for detailed explanations.

Any comments considered necessary by applicant must be submitted no later than the payment of the issue fee and, to avoid processing delays, should preferably accompany the issue fee. Such submissions should be clearly labeled "Comments on Statement of Reasons for Allowance."

3. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Michael Opsasnick, telephone number (703)305-4089.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Mr. Fan Tsang, can be reached at (703)305-4895. The facsimile phone number for this group is (703)308-6306.

Any inquiry of a general nature or relating to the status of this applications should be directed to the Group receptionist whose telephone number is (703)305-3900.

September 25, 2001

mno

FAN TSANG
SUPERVISORY PATENT EXAMINER
TECHNOLOGY CENTER 2600

UNITED STATES DEPARTMENT OF COMMERCE
Patent and Trademark Office

NOTICE OF ALLOWANCE AND ISSUE FEE DUE

WM02/0928
BEVERIDGE, DEGRANDI, WEILACHER & YOUNG
1850 M STREET, N. W., SUITE 800
WASHINGTON DC 20036

APPLICATION NO.	FILING DATE	TOTAL CLAIMS	EXAMINER AND GROUP ART UNIT	DATE MAILED
08/161,798	12/06/93	020	OPSASNICK, M 2645	09/28/01
First Named Applicant: TEH.			35 USC 154(b) term ext. = 0 Days.	

TITLE OF INVENTION: BIT ALLOCATION METHOD FOR DIGITAL AUDIO SIGNALS

ATTY'S DOCKET NO.	CLASS-SUBCLASS	BATCH NO.	APPLN. TYPE	SMALL ENTITY	FEE DUE	DATE DUE
2 0348683	704-229.000	M73	UTILITY	NO	$1240.00	12/28/01

THE APPLICATION IDENTIFIED ABOVE HAS BEEN EXAMINED AND IS ALLOWED FOR ISSUANCE AS A PATENT. PROSECUTION ON THE MERITS IS CLOSED.

THE ISSUE FEE MUST BE PAID WITHIN THREE MONTHS FROM THE MAILING DATE OF THIS NOTICE OR THIS APPLICATION SHALL BE REGARDED AS ABANDONED. THIS STATUTORY PERIOD CANNOT BE EXTENDED.

HOW TO RESPOND TO THIS NOTICE:

I. Review the SMALL ENTITY status shown above.
If the SMALL ENTITY is shown as YES, verify your current SMALL ENTITY status:

A. If the status is changed, pay twice the amount of the FEE DUE shown above and notify the Patent and Trademark Office of the change in status, or
B. If the status is the same, pay the FEE DUE shown above.

If the SMALL ENTITY is shown as NO:

A. Pay FEE DUE shown above, or

B. File verified statement of Small Entity Status before, or with, payment of 1/2 the FEE DUE shown above.

II. Part B-Issue Fee Transmittal should be completed and returned to the Patent and Trademark Office (PTO) with your ISSUE FEE. Even if the ISSUE FEE has already been paid by charge to deposit account, Part B Issue Fee Transmittal should be completed and returned. If you are charging the ISSUE FEE to your deposit account, section "4b" of Part B-Issue Fee Transmittal should be completed and an extra copy of the form should be submitted.

III. All communications regarding this application must give application number and batch number.
Please direct all communications prior to issuance to Box ISSUE FEE unless advised to the contrary.

IMPORTANT REMINDER: Utility patents issuing on applications filed on or after Dec. 12, 1980 may require payment of maintenance fees. It is patentee's responsibility to ensure timely payment of maintenance fees when due.

PATENT AND TRADEMARK OFFICE COPY

PTOL-85 (REV. 10-96) Approved for use through 06/30/99. (0651-0033)

BEST COPY

PART B—ISSUE FEE TRANSMITTAL

Complete and mail this form, together with app. [illegible] fees, to: Box ISSUE FEE
Assistant Commissioner for Patents
Washington, D.C. 20231

12/28/01

DEC 03 2001

48683 B8

MAILING INSTRUCTIONS: This form should be used for transmitting the ISSUE FEE. Blocks 1 through 4 should be completed where appropriate. All further correspondence including the Issue Fee Receipt, the Patent, advance orders and notification of maintenance fees will be mailed to the current correspondence address as indicated unless corrected below or directed otherwise in Block 1, by (a) specifying a new correspondence address; and/or (b) indicating a separate "FEE ADDRESS" for maintenance fee notifications.

CURRENT CORRESPONDENCE ADDRESS (Note: Legibly mark-up with any corrections or use Block 1)

WM02/0928
BEVERIDGE, DEGRANDI, WEILACHER & YOUNG
1850 M STREET, N. W., SUITE 800
WASHINGTON DC 20036

Note: The certificate of mailing below can only be used for domestic mailings of the Issue Fee Transmittal. This certificate cannot be used for any other accompanying papers. Each additional paper, such as an assignment or formal drawing, must have its own certificate of mailing.

Certificate of Mailing

I hereby certify that this Issue Fee Transmittal is being deposited with the United States Postal Service with sufficient postage for first class mail in an envelope addressed to the Box Issue Fee address above on the date indicated below.

(Depositor's name)

(Signature)

(Date)

APPLICATION NO.	FILING DATE	TOTAL CLAIMS	EXAMINER AND GROUP ART UNIT	DATE MAILED
08/161,798	12/06/93	020	OPSASNICK, M 2645	09/28/01

First Named Applicant: TEH, 35 USC 154(b) term ext. = 0 Days.

TITLE OF INVENTION: BIT ALLOCATION METHOD FOR DIGITAL AUDIO SIGNALS

ATTY'S DOCKET NO.	CLASS-SUBCLASS	BATCH NO.	APPLN. TYPE	SMALL ENTITY	FEE DUE	DATE DUE
2 0348683	704-229.000	M73	UTILITY	NO	~~$1240.00~~ $1280.00	12/28/01

1. Change of correspondence address or indication of "Fee Address" (37 CFR 1.363). Use of PTO form(s) and Customer Number are recommended, but not required.

☐ Change of correspondence address (or Change of Correspondence Address form PTO/SB/122) attached.

☐ "Fee Address" indication (or "Fee Address" Indication form PTO/SB/47) attached.

2. For printing on the patent front page, list (1) the names of up to 3 registered patent attorneys or agents OR, alternatively, (2) the name of a single firm (having as a member a registered attorney or agent) and the names of up to 2 registered patent attorneys or agents. If no name is listed, no name will be printed.

1 Smith, Gambrell & Russell, LLP
2
3

3. ASSIGNEE NAME AND RESIDENCE DATA TO BE PRINTED ON THE PATENT (print or type)
PLEASE NOTE: Unless an assignee is identified below, no assignee data will appear on the patent. Inclusion of assignee data is only appropriate when an assignment has been previously submitted to the PTO or is being submitted under separate cover. Completion of this form is NOT a substitute for filing an assignment.

(A) NAME OF ASSIGNEE : Matsushita Electric Industrial Co., Ltd.

(B) RESIDENCE: (CITY & STATE OR COUNTRY) : Osaka, JAPAN

Please check the appropriate assignee category indicated below (will not be printed on the patent)
☐ individual X corporation or other private group entity ☐ government

4a. The following fees are enclosed (make check payable to Commissioner of Patents and Trademarks):
X Issue Fee
☐ Advance Order - # of Copies

4b. The following fees or deficiency in these fees should be charged to:
DEPOSIT ACCOUNT NUMBER 02-4300
(ENCLOSE AN EXTRA COPY OF THIS FORM)
X Issue Fee
☐ Advance Order - # of Copies

The COMMISSIONER OF PATENTS AND TRADEMARKS is requested to apply the Issue Fee to the application identified above.

(Authorized Signature) (Date)
Michael A. Makuch, Reg. 32,263 Dec 3 2001

NOTE: The Issue Fee will not be accepted from anyone other than the applicant; a registered attorney or agent; or the assignee or other party in interest as shown by the records of the Patent and Trademark Office.

12/05/2001 MFANAEI1 00000063 08161798
01 FC:142 1280.00 OP

Burden Hour Statement: This form is estimated to take 0.2 hours to complete. Time will vary depending on the needs of the individual case. Any comments on the amount of time required to complete this form should be sent to the Chief Information Officer, Patent and Trademark Office, Washington, D.C. 20231. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. SEND FEES AND THIS FORM TO: Box Issue Fee, Assistant Commissioner for Patents, Washington D.C. 20231

Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it displays a valid OMB control number.

TRANSMIT THIS FORM WITH FEE

#2645

Attorney Docket
32705 M 48683



PATENT

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicant : Do Hui TEH, et al.
Serial No. : 08/161,798 Art Unit : 2645
Filed : December 6, 1993 Examiner : M. Opsasnick
For : Bit Allocation Method for Digital Audio Signals

--- BATCH M 73 ---

FORMAL DRAWING TRANSMITTAL LETTER

Commissioner of Patents
Washington, D.C. 20231

Attn: Official Draftsperson

Sir:

The Issue Fee for the above-referenced patent application, due December 28, 2001, is being paid concurrent herewith.

In accordance with the corrections required by the Notice of Draftsperson's Patent Drawing Review (attached to Paper No.5), attached are eight sheets of corrected formal drawings, showing Figures 1-8, to replace the drawings first filed with the application on December 6, 1993.

Respectfully submitted,

SMITH, GAMBRELL & RUSSELL, LLP

By: _____

Michael A. Makuch, Reg. 32,263
1850 M Street, NW – Suite 800
Washington, DC 20036
Telephone : 202/263-4300
Facsimile : 202/263-4329

December 3, 2001

170680w

9/01

Fig. 1

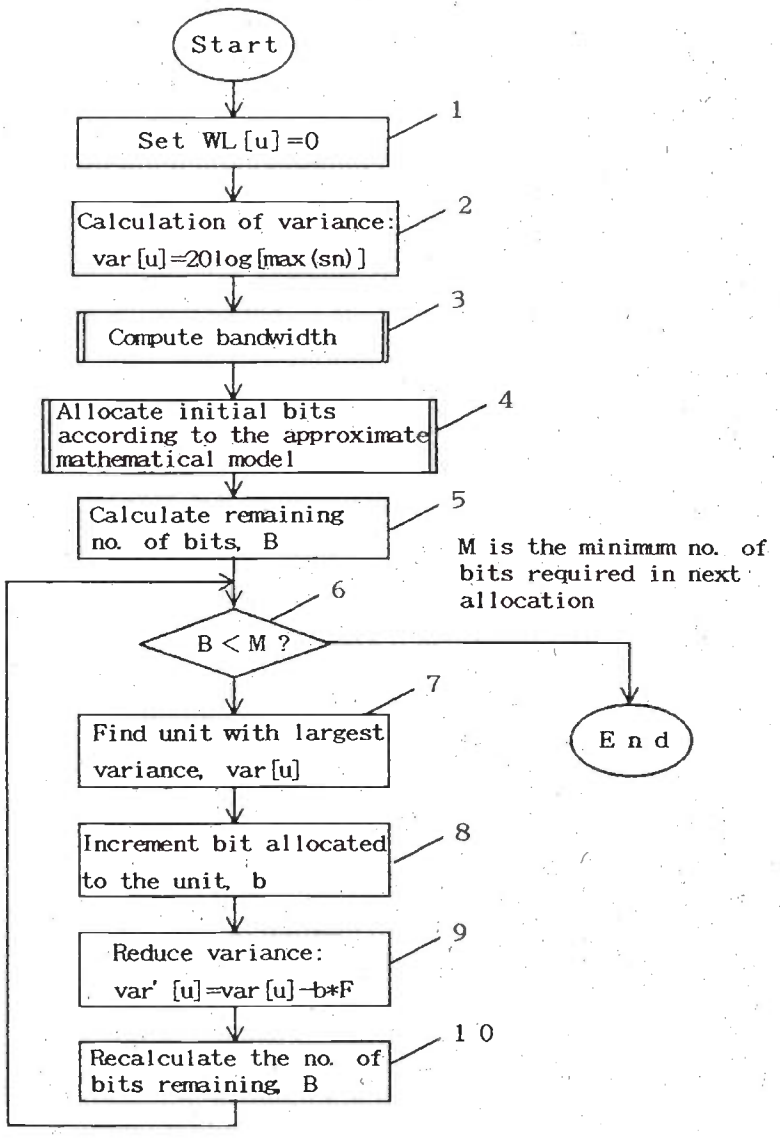

Start
Set WL[u]=0
1
Calculation of variance:
var[u]=20log[max(sn)]
2
Compute bandwidth
3
Allocate initial bits according to the approximate mathematical model
4
Calculate remaining no. of bits, B
5
M is the minimum no. of bits required in next allocation
6
B < M ?
End
7
Find unit with largest variance, var[u]
Increment bit allocated to the unit, b
8
Reduce variance:
var'[u]=var[u]-b*F
9
Recalculate the no. of bits remaining, B
10

Fig. 2

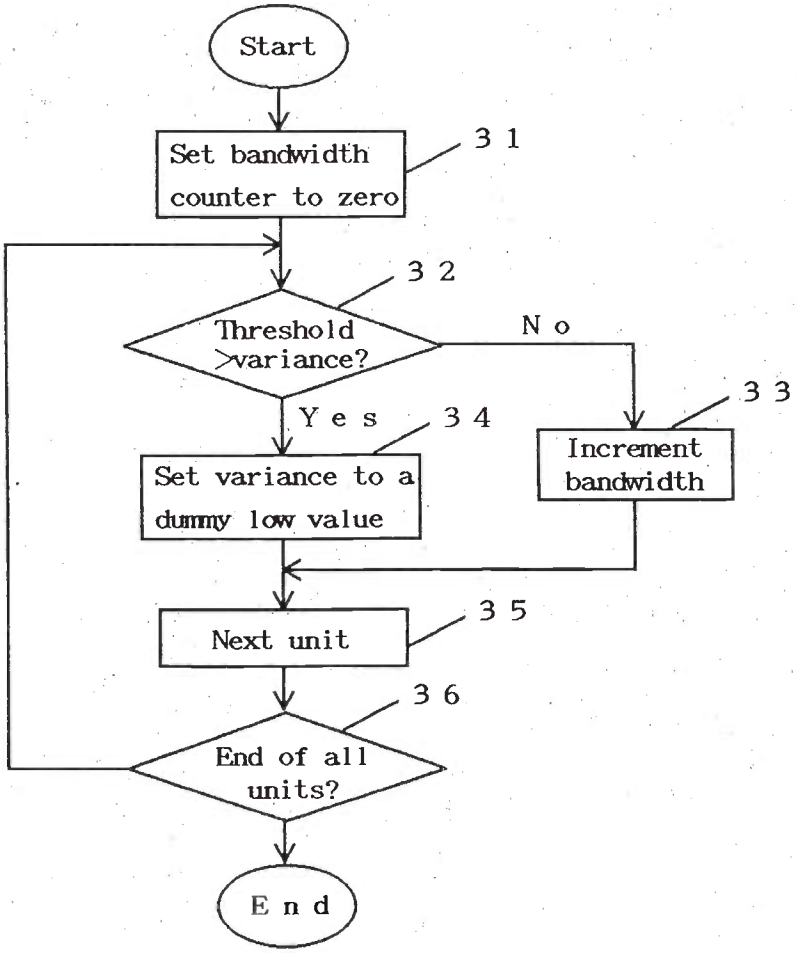

Start
Set bandwidth counter to zero
31
32
Threshold >variance?
No
Yes
33
34
Increment bandwidth
Set variance to a dummy low value
Next unit
35
36
End of all units?
End

Fig. 3

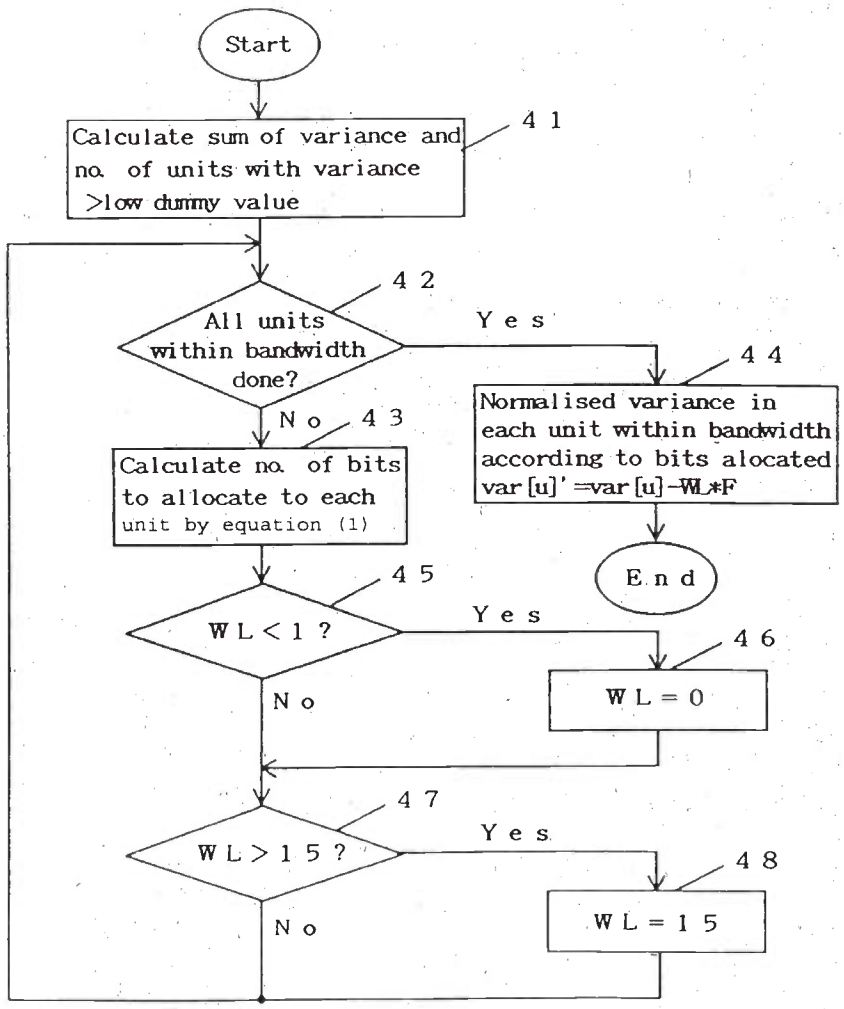
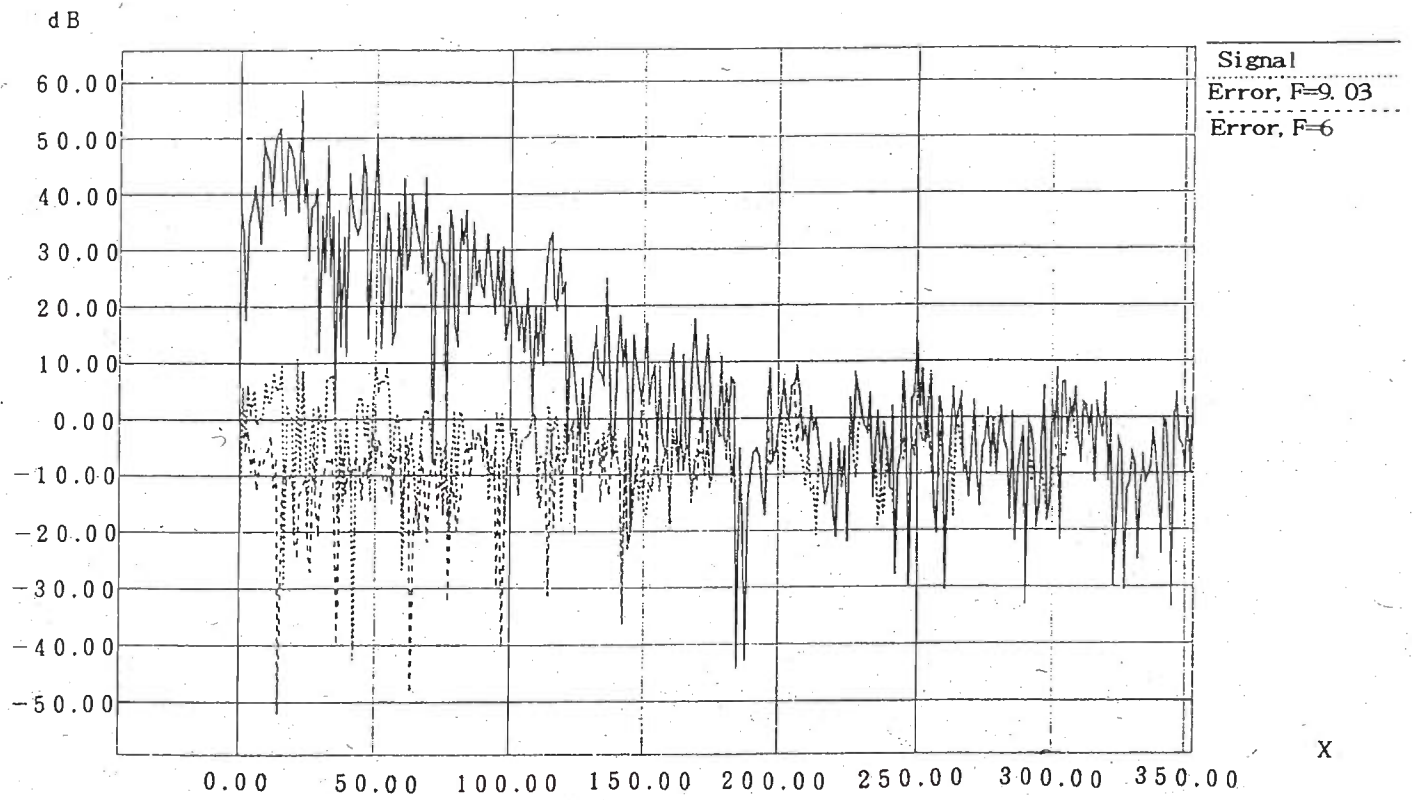

Start
Calculate sum of variance and no. of units with variance >low dummy value
41
All units within bandwidth done?
42
Yes
No
43
Calculate no. of bits to allocate to each unit by equation (1)
44
Normalised variance in each unit within bandwidth according to bits alocated var[u]' =var[u]-WL*F
End
WL < 1 ?
45
Yes
No
46
WL = 0
WL > 15 ?
47
Yes
No
48
WL = 15

Fig. 4


dB
60.00
50.00
40.00
30.00
20.00
10.00
0.00
-10.00
-20.00
-30.00
-40.00
-50.00
0.00
50.00
100.00
150.00
200.00
250.00
300.00
350.00
X
Signal
Error, F=9.03
Error, F=6

0273

Fig. 5

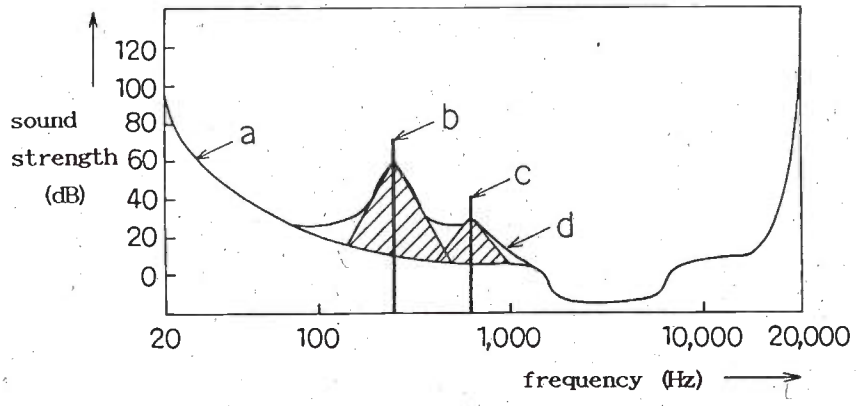

120
100
80
60
40
20
0
sound
strength
(dB)
a
b
c
d
20
100
1,000
10,000
20,000
frequency (Hz)

Fig. 6

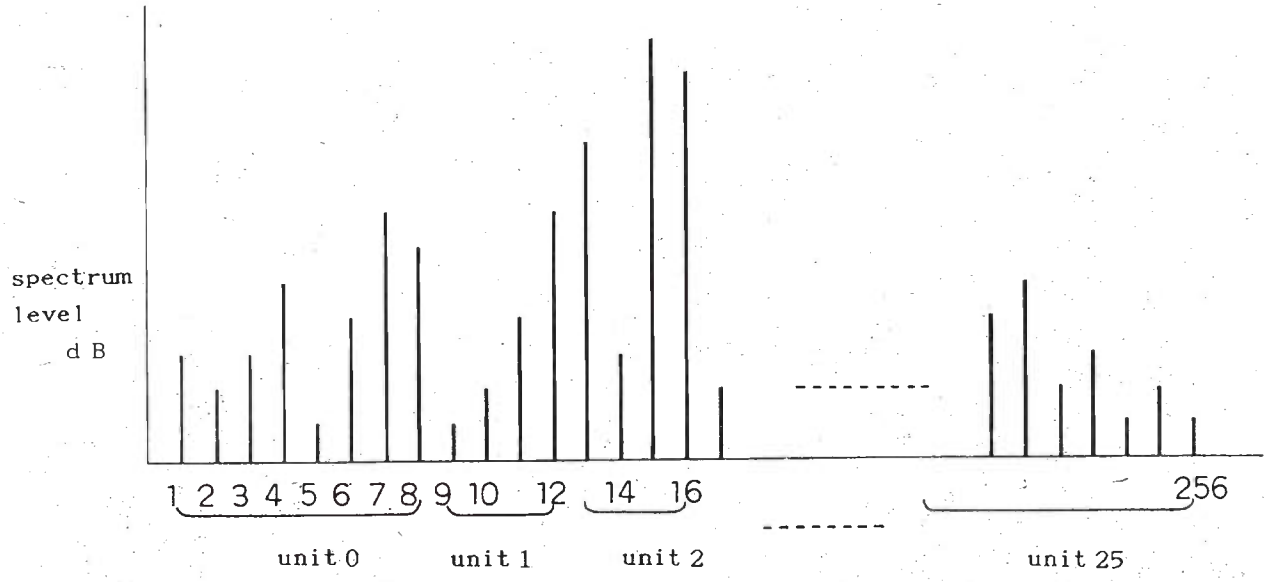
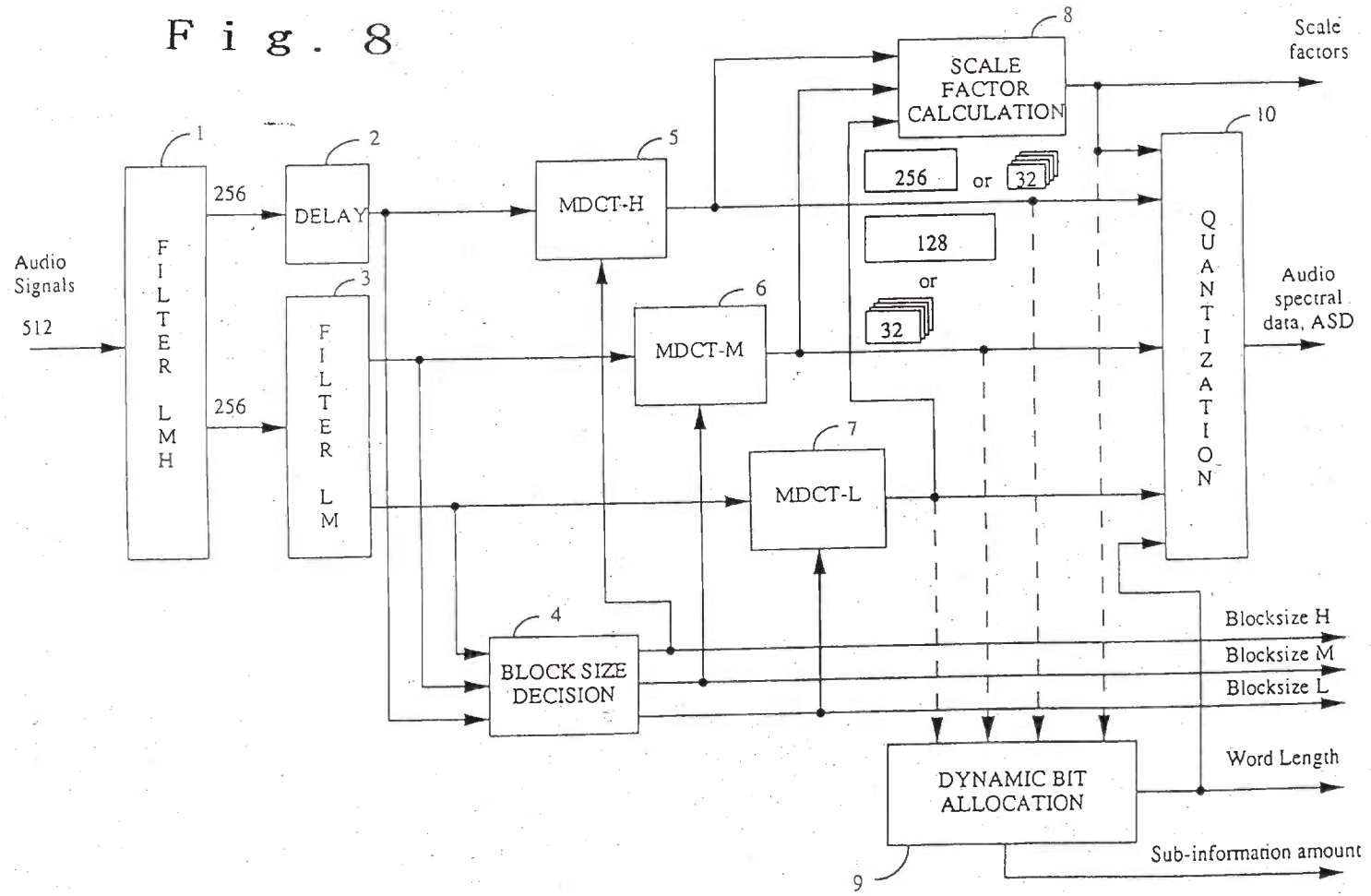

spectrum
level
d B
1 2 3 4 5 6 7 8 9 10 12 14 16 256
unit 0
unit 1
unit 2
unit 25

Fig. 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)

Fig. 8


Audio Signals
512
1
FILTER LMH
256
256
2
DELAY
3
FILTER LM
4
BLOCK SIZE DECISION
5
MDCT-H
6
MDCT-M
7
MDCT-L
8
SCALE FACTOR CALCULATION
256
or
32
128
or
32
9
DYNAMIC BIT ALLOCATION
10
QUANTIZATION
Scale factors
Audio spectral data, ASD
Blocksize H
Blocksize M
Blocksize L
Word Length
Sub-information amount

0277


OIPE JC87
DEC 0 3 2001
PATENT & TRADEMARK OFFICE

File History Content Report

The following content is missing from the original file history record obtained from the United States Patent and Trademark Office. No additional information is available.

Document Date - 2002-01-15

Document Title - USPTO Grant

This page is not part of the official USPTO record. It has been determined that content identified on this document is missing from the original file history record.

OIPE IAP45 SEP 12 2005 PATENT & TRADEMARK OFFICE

Attorney Docket No. 24119

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re Application of:

Larry Hinderks

Serial No. 09/664,727

Filed: September 19, 2000

For: **AUDIO FILE DISTRIBUTION AND PRODUCTION SYSTEM**

TRANSMITTAL LETTER

Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313

Sir:

Submitted herewith for filing in the U.S. Patent and Trademark Office is the following:

(1) Transmittal Letter; and
(2) Request for Withdrawl as Attorney or Agent and Change of Correspondence Address.

Respectfully submitted,
NATH & ASSOCIATES PLLC

September 12, 2005

NATH & ASSOCIATES PLLC
1030 15th Street, N.W.
6th Floor
Washington, D.C. 20005
Tel: (202) 775-8383
Fax: (202) 775-8396

Gary M. Nath
Reg. No. 26,965
Gregory B. Kang
Reg. No. 45,273
Customer No. 20529

6339756

OCS

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SEP 12 2005
PATENT & TRADEMARK OFFICE

COMPLETED

PTO/SB/83 (04-05)
Approved for use through 11/30/2005. OMB 0651-0035
U.S. Patent and Trademark Office, U.S. DEPARTMENT OF COMMERCE
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REQUEST FOR WITHDRAWAL AS ATTORNEY OR AGENT AND CHANGE OF CORRESPONDENCE ADDRESS

Application Number	09/664,727
Filing Date	09-19-2000
First Named Inventor	Larry Hinderks
Art Unit	2645
Examiner Name	Knepper, David D
Attorney Docket Number	24119

To: Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Please withdraw me as attorney or agent for the above identified patent application, and

☐ all the attorneys/agents of record.

☐ the attorneys/agents (with registration numbers) listed on the attached paper(s), or

☒ the attorneys/agents associated with Customer Number 20529

NOTE: This box can only be checked when the power of attorney of record in the application is to all the practitioners associated with a customer number.

The reasons for this request are:

the client requested that we not further prosecute the application.

CORRESPONDENCE ADDRESS

1. ☐ The correspondence address is NOT affected by this withdrawal.
2. ☒ Change the correspondence address and direct all future correspondence to:

☐ The address associated with Customer Number:

OR

☒ Firm or Individual Name	STARGUIDE DIGITAL NETWORKS, INC., Elaine Locke		
Address	750 West John Carpenter Freeway Suite #700		
City	Irving	State TX	Zip 75039
Country	USA		
Telephone	(972)581-2000	Email	hq@starguidedigital.com
Signature			
Name	Gregory B. Kang	Registration No.	45,273
Date		Telephone No.	(202)775-8383

NOTE: Withdrawal is effective when approved rather than when received. Unless there are at least 30 days between approval of withdrawal and the expiration date of a time period for response or possible extension period, the request to withdraw is normally disapproved.

This collection of information is required by 37 CFR 1.36. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.11 and 1.14. This collection is estimated to take 12 minutes to complete, including gathering, preparing, and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. SEND TO: **Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

If you need assistance in completing the form, call 1-800-PTO-9199 and select option 2.

UNITED STATES PATENT AND TRADEMARK OFFICE

UNITED STATES DEPARTMENT OF COMMERCE
United States Patent and Trademark Office
Address: COMMISSIONER FOR PATENTS
P.O. Box 1450
Alexandria, Virginia 22313-1450
www.uspto.gov

APPLICATION NUMBER	FILING OR 371 (c) DATE	FIRST NAMED APPLICANT	ATTY. DOCKET NO./TITLE
09/664,727	09/19/2000	Larry W. Hinderks	24119

CONFIRMATION NO. 1468

OC0000000177518577

20529
NATH & ASSOCIATES
112 South West Street
Alexandria, VA 22314

Date Mailed: 12/30/2005

NOTICE REGARDING CHANGE OF POWER OF ATTORNEY

This is in response to the Power of Attorney filed 09/12/2005.

- The withdrawal as attorney in this application has been accepted. Future correspondence will be mailed to the new address of record. 37 CFR 1.33.

ANITA C GREENE
OIPE (703) 308-9010

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www.uspto.gov

APPLICATION NUMBER	FILING OR 371 (c) DATE	FIRST NAMED APPLICANT	ATTY. DOCKET NO./TITLE
09/664,727	09/19/2000	Larry W. Hinderks	24119

CONFIRMATION NO. 1468

OC000000017751905

STARGUIDE DIGITAL NETWORKS INC
ELAINE LOCKE
750 WEST JOHN CARPENTER FREEWAY
SUITE 700
IRVING, TX 75039

Date Mailed: 12/30/2005

NOTICE OF ACCEPTANCE OF POWER OF ATTORNEY

This is in response to the Power of Attorney filed 09/12/2005.

The Power of Attorney in this application is accepted. Correspondence in this application will be mailed to the above address as provided by 37 CFR 1.33.

ANITA C GREENE
OIPE (703) 308-9010

OFFICE COPY

PATENT APPLICATION FEE DETERMINATION RECORD

Effective October 1, 1992

Application or Docket Number: 161798

CLAIMS AS FILED - PART I

FOR	(Column 1) NUMBER FILED	(Column 2) NUMBER EXTRA	SMALL ENTITY RATE	SMALL ENTITY FEE	OR	OTHER THAN SMALL ENTITY RATE	OTHER THAN SMALL ENTITY FEE
BASIC FEE				$355.00	OR		$710.00
TOTAL CLAIMS	4 minus 20 =	*	x$11=		OR	x$22=	
INDEPENDENT CLAIMS	1 minus 3 =	*	x 37=		OR	x 74=	
MULTIPLE DEPENDENT CLAIM PRESENT			+115=		OR	+230=	
* If the difference in column 1 is less then zero, enter "0" in column 2			TOTAL		OR	TOTAL	710

CLAIMS AS AMENDED - PART II

AMENDMENT A

	(Column 1) CLAIMS REMAINING AFTER AMENDMENT		(Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR	(Column 3) PRESENT EXTRA	SMALL ENTITY RATE	SMALL ENTITY ADDITIONAL FEE	OR	OTHER THAN SMALL ENTITY RATE	OTHER THAN SMALL ENTITY ADDITIONAL FEE
Total	* 21	Minus	** 20	= 1	x$11=		OR	x$22=	22.00
Independent	* 1	Minus	*** 3	=	x 37=		OR	x 74=	
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM					+ 115=		OR	+230=	
					TOTAL ADDIT. FEE		OR	TOTAL ADDIT. FEE	2200 Paid

AMENDMENT B

	(Column 1) CLAIMS REMAINING AFTER AMENDMENT		(Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR	(Column 3) PRESENT EXTRA	SMALL ENTITY RATE	SMALL ENTITY ADDITIONAL FEE	OR	OTHER THAN SMALL ENTITY RATE	OTHER THAN SMALL ENTITY ADDITIONAL FEE
Total	*	Minus	**	=	x$11=		OR	x$22=	
Independent	*	Minus	***	=	x 37=		OR	x 74=	
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM					+ 115=		OR	+ 230=	
					TOTAL ADDIT. FEE		OR	TOTAL ADDIT. FEE	

AMENDMENT C

	(Column 1) CLAIMS REMAINING AFTER AMENDMENT		(Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR	(Column 3) PRESENT EXTRA	SMALL ENTITY RATE	SMALL ENTITY ADDITIONAL FEE	OR	OTHER THAN SMALL ENTITY RATE	OTHER THAN SMALL ENTITY ADDITIONAL FEE
Total	*	Minus	**	=	x$11=		OR	x$22=	
Independent	*	Minus	***	=	x 37=		OR	x 74=	
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM					+115=		OR	+230=	
					TOTAL ADDIT. FEE		OR	TOTAL ADDIT. FEE	

* If the entry in column 1 is less than the entry in column 2, write "0" in column 3.
** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 20, enter "20".
*** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 3, enter "3".
The "Highest Number Previously Paid For" (Total or Independent) is the highest number found in the appropriate box in column 1.

FORM PTO-875
(Rev.10-92)

Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE

PTO 1130 (REV 11/91)

U.S. DEPARTMENT OF COMMERCE– PATENT & TRADEMARK OFFICE

PACE DATA ENTRY CODING SHEET

1ST EXAMINER PHACKLEY — DATE 12.27.93

2ND EXAMINER — DATE

APPLICATION NUMBER	TYPE APPL	FILING DATE MONTH	DAY	YEAR	SPECIAL HANDLING	GROUP ART UNIT	CLASS	SHEETS OF DRAWING
08/161798	1	12	06	93		2104	341	8

TOTAL CLAIMS	INDEPENDENT CLAIMS	SMALL ENTITY?	FILING FEE	FOREIGN LICENSE	ATTORNEY DOCKET NUMBER
4	1		710	Y	034868

CONTINUITY DATA

CONTINUITY CODE	STATUS CODE	PARENT APPPLICATION SERIAL NUMBER	PARENT PATENT NUMBER	PARENT FILING DATE MONTH	DAY	YEAR
		0				
		0				
		0				
		0				
		0				

PCT/FOREIGN APPLICATION DATA

FOREIGN PRIORITY CLAIMED	COUNTRY CODE	PCT/FOREIGN APPLICATION SERIAL NUMBER	FOREIGN FILING DATE MONTH	DAY	YEAR
Y	JPX	5-03712	02	19	93

0285

Derwent Innovation

Derwent Innovation Patent Export, 2021-09-16 00:42:30 +0000

Search results for: PNS=(US6339757);
Collections searched: DWPI; ARIPO grant; Argentina UM; Argentina app; Armenia grant; Australia UM; Australia app; Australia grant; Austria UM; Austria app; Austria grant; Belarus UM; Belarus grant; Belgium app; Belgium grant; Brazil UM; Brazil app; Brazil grant; Bulgaria UM; Bulgaria app; Bulgaria grant; Canada app; Canada grant; China, Mainland UM; China, Mainland app; China, Mainland grant; Colombia app; Costa Rica UM; Costa Rica app; Croatia UM; Croatia app; Croatia grant; Cuba app; Cuba grant; Czech Republic UM; Czech Republic app; Czech Republic grant; Czechoslovakia grant; Denmark UM; Denmark app; Denmark grant; Estonia UM; Estonia app; Estonia grant; Eurasian (EAPO) grant; Europe (EP) app; Europe (EP) grant; Finland app; Finland grant; France app; France grant; Georgia grant; German Democratic Republic app; German Democratic Republic grant; Germany UM; Germany app; Germany grant; Greece app; Greece grant; Gulf Cooperation Council grant; Hong Kong UM; Hong Kong app; Hong Kong grant; Hungary grant; Iceland grant; India app; India grant; Indonesia UM; Indonesia app; Ireland UM; Ireland app; Ireland grant; Israel grant; Italy grant; Italy grant; Italy grant; Italy grant; Japan UM; Japan app; Japan grant; Korea UM; Korea app; Korea grant; Latvia grant; Lithuania grant; Luxembourgh app; Luxembourgh grant; Malaysia grant; Mexico app; Mexico grant; Moldova UM; Moldova grant; Monaco grant; Mongolia UM; Mongolia grant; Morocco app; Morocco grant; Netherlands app; Netherlands grant; New Zealand grant; Norway app; Norway grant; OAPI grant; PCT (WO) app; Philippines app; Philippines grant; Poland UM; Poland grant; Portugal UM; Portugal app; Portugal grant; Romania UM; Romania app; Romania granted; Russia UM; Russia app; Russia grant; Serbia UM; Serbia app; Serbia grant; Singapore app; Singapore grant; Slovakia UM; Slovakia app; Slovakia grant; Slovenia UM; Slovenia grant; South Africa app; Spain UM; Spain app; Spain grant; Sweden app; Sweden grant; Switzerland app; Switzerland grant; Taiwan UM; Taiwan appl; Taiwan grant; Thailand grant; Tunisia app; Turkey UM; Turkey app; UK app; UK grant; USSR grant; Ukraine UM; Ukraine grant; United States app; United States grant; Uruguay app; Vietnam app; Vietnam grant; Other

Table of Contents

1. US6339757B1 Bit allocation method for digital audio signals

Family 1/1

1 record(s) per family

Record 1/1 US6339757B1 Bit allocation method for digital audio signals

Title: Bit allocation method for digital audio signals

Title - DWPI: Determining quantisation of digital audio signals for bit allocation for transform coder adapting dynamic bit allocation to human auditory system by computing necessary bandwidth of components of transmission frame using psycho-acoustics criteria

Priority Number: JP199330712A

Priority Date: 1993-02-19

Application Number: US1993161798A

Application Date: 1993-12-06

Publication Date: 2002-01-15

IPC Class Table:

IPC	Section	Class	Subclass	Class Group	Subgroup
G10L001900	G	G10	G10L	G10L0019	G10L001900
G10L001902	G	G10	G10L	G10L0019	G10L001902
H03M000730	H	H03	H03M	H03M0007	H03M000730
H03M000750	H	H03	H03M	H03M0007	H03M000750
H04B000166	H	H04	H04B	H04B0001	H04B000166

IPC Class Table - DWPI:

IPC - DWPI	Section - DWPI	Class - DWPI	Subclass - DWPI	Class Group - DWPI	Subgroup - DWPI
G10L001900 (IPC 1-7)	G	G10	G10L	G10L0019	G10L001900 (IPC 1-7)
G10L001904 (IPC 1-7)	G	G10	G10L	G10L0019	G10L001904 (IPC 1-7)
G10L000918 (IPC 1-7)	G	G10	G10L	G10L0009	G10L000918 (IPC 1-7)
H04B000166 (IPC 1-7)	H	H04	H04B	H04B0001	H04B000166 (IPC 1-7)
G10L001902 (IPC	G	G10	G10L	G10L0019	G10L001902 (IPC

1-7)					1-7)
H03M000750 (IPC 1-7)	H	H03	H03M	H03M0007	H03M000750 (IPC 1-7)
G10L001900	G	G10	G10L	G10L0019	G10L001900
G10L001902	G	G10	G10L	G10L0019	G10L001902
H03M000730	H	H03	H03M	H03M0007	H03M000730
H03M000750	H	H03	H03M	H03M0007	H03M000750
H04B000166	H	H04	H04B	H04B0001	H04B000166

Assignee/Applicant: Matsushita Electric Industrial Co. Ltd.,Osaka,JP

JP F Terms:

JP FI Codes:

Assignee - Original: Matsushita Electric Industrial Co. Ltd.

Any CPC Table:

Type	Invention	Additional	Version	Office
Current	**H04B 1/665**	-	20130101	EP; US
Original	**H04B 1/665**	-	20130101	US

ECLA: H04B000166M

Abstract:

When determining a quantization for digital audio signals having a spectral and temporal structure wherein the audio signals are buffered in frames and decomposed into spectral components, the method is: obtaining a variance and/or a representative of the components of the the frame within a defined frequency interval; computing a necessary bandwidth using psychoacoustic criteria; determining an initial quantization for each frequency interval using an approximate mathematical model; and iteratively increasing the quantization of each of the frequency intervals through a psychoacoustically weighted inverse relationship with the the variance of each of the frequency intervals.

Language of Publication: EN

INPADOC Legal Status Table:

Gazette Date	Code	INPADOC Legal Status Impact

2013-03-14	FPAY	+
Description: FEE PAYMENT FEE PAYMENT YEAR 12		
2013-01-18	AS	-
Description: ASSIGNMENT DOLBY INTERNATIONAL AB, NETHERLANDS ASSIGNMENT OF ASSIGNORS INTEREST; ASSIGNOR:PANASONIC CORPORATION; REEL/FRAME:029654/0754 2012-10-30		
2012-11-12	AS	-
Description: ASSIGNMENT PANASONIC CORPORATION, JAPAN CHANGE OF NAME; ASSIGNOR:MATSUSHITA ELECTRIC INDUSTRIAL CO., LTD.; REEL/FRAME:029283/0355 2008-10-01		
2009-06-17	FPAY	+
Description: FEE PAYMENT FEE PAYMENT YEAR 8		
2005-06-21	FPAY	+
Description: FEE PAYMENT FEE PAYMENT YEAR 4		
2001-12-28	STCF	-
Description: INFORMATION ON STATUS: PATENT GRANT PATENTED CASE		
1997-11-10	AS	-
Description: ASSIGNMENT MATSUSHITA ELECTRIC INDUSTRIAL CO., LTD., JAPAN RE-RECORDED TO CORRECT THE ADDRESS OF THE ASSIGNEE; ASSIGNORS:TEH, DO HUI; TAN, PEK YEW; NEO, SUA HONG; REEL/FRAME:008787/0297 1993-10-04		
1997-09-12	AS	-
Description: ASSIGNMENT MATSUSHITA ELECTRIC INDUSTRIAL CO., LTD., JAPAN CORRECTION OF NOTICE OF RECORDATION OF ASSIGNMENT RECORDED ON 12-6-1993 AT REEL/FRAME 6798/0252 DUE TO APPLICANTS ERROR. PLEASE CHANGE THE NAME OF THE ASSIGNEE FROM "MASTSUSHITA ELECTRIC INDUSTRIAL CO., LTD." TO MATSUSHITA ELECTRIC INDUSTRIAL CO., LTD.; ASSIGNORS:TEH, DO HUI; TAN, PEK YEW; NEO, SUA HONG; REEL/FRAME:008697/0210 1993-10-04		
1993-12-06	AS	-
Description: ASSIGNMENT MASTSUSHITA ELECTRIC INDUSTRIAL CO., LTD. A COR ASSIGNMENT OF ASSIGNORS INTEREST; ASSIGNORS:TEH, DO HUI; TAN, PEK YEW; NEO, SUA HONG; REEL/FRAME:006798/0252 1993-10-04		

Post-Issuance (US):

Reassignment (US) Table:

Assignee	Assignor	Date Signed	Reel/Frame	Date
DOLBY INTERNATIONAL AB,AMSTERDAM ZUIDOOST,NL	PANASONIC CORPORATION	2012-10-30	029654/0754	2013-01-18
Conveyance: ASSIGNMENT OF ASSIGNORS INTEREST (SEE DOCUMENT FOR DETAILS).				
Corresponent: BIANCA MIYAKAWA 475 BRANNAN STREET, SUITE 300 SAN FRANCISCO, CA 94107-5420				
PANASONIC CORPORATION,OSAKA,JP	MATSUSHITA ELECTRIC INDUSTRIAL CO., LTD.	2008-10-01	029283/0355	2012-11-12
Conveyance: CHANGE OF NAME (SEE DOCUMENT FOR DETAILS).				
Corresponent: PANASONIC PATENT CENTER 1130 CONNECTICUT AVE., N.W., SUITE 1100 WASHINGTON, DC 20036				
MATSUSHITA ELECTRIC INDUSTRIAL CO. LTD.,OSAKA 571,JP	TEH, DO HUI	1993-10-04	008787/0297	1997-11-10
	TAN, PEK YEW	1993-10-04	-	-
	NEO, SUA HONG	1993-10-04	-	-
Conveyance: RE-RECORDED TO CORRECT THE ADDRESS OF THE ASSIGNEE				
Corresponent: DEGRANDI,WEILACHER & YOUNG, LLP JOSEPH A. DEGRANDI 1850 M. STREET, N.W. SUITE 800 WASHINGTON, D.C. 20036				
MATSUSHITA ELECTRIC INDUSTRIAL CO. LTD.,OSAKA 571,JP	TEH, DO HUI	1993-10-04	008697/0210	1997-09-12
	TAN, PEK YEW	1993-10-04	-	-
	NEO, SUA HONG	1993-10-04	-	-
Conveyance: CORRECTION OF NOTICE OF RECORDATION OF ASSIGNMENT RECORDED ON 12-6-1993 AT REEL/FRAME 6798/0252 DUE TO APPLICANTS ERROR. PLEASE CHANGE THE NAME OF THE ASSIGNEE FROM "MASTSUSHITA ELECTRIC INDUSTRIAL CO., LTD." TO MATSUSHITA ELECTRIC INDUSTRIAL CO., LTD.				
Corresponent: BEVERIDGE, DEGRANDI, WEILACHER, ET AL. JOSEPH A. DEGRANDI 1850 M STREET N.W. SUITE 800 WASHINGTON, D.C. 20036				
MASTSUSHITA ELECTRIC INDUSTRIAL CO. LTD. A CORPORATION OF JAPAN,OSAKA,JP	TEH, DO HUI	1993-10-04	006798/0252	1993-12-06
	TAN, PEK YEW	1993-10-04	-	-
	NEO, SUA HONG	1993-10-04	-	-
Conveyance: ASSIGNMENT OF ASSIGNORS INTEREST (SEE DOCUMENT FOR DETAILS).				
Corresponent: BEVERIDGE, DEGRANDI, WEILACHER & YOUNG JOSEPH A. DEGRANDI 1850 M ST., NW SUITE 800 WASHINGTON, DC 20036				

Maintenance Status (US):

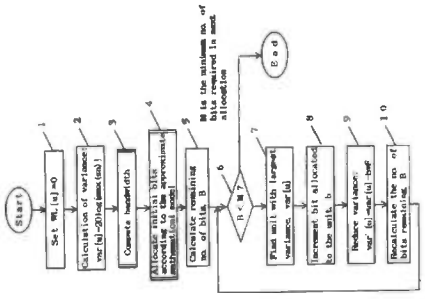
Litigation (US):

Opposition (EP):

License (EP):

EPO Procedural Status:

Front Page Drawing:



Priority Number - DWPI: JP199330712A
Priority Date - DWPI: 1993-02-19
Assignee - Current US: DOLBY INTERNATIONAL AB

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United States Patent and Trademark Office

Office of the Commissioner for Patents

BIT ALLOCATION METHOD FOR DIGITAL AUDIO SIGNALS

PATENT #	APPLICATION #	FILING DATE	ISSUE DATE
6339757	08161798	12/06/1993	01/15/2002

Payment Window Status

No maintenance fees are due.

WINDOW	STATUS	FEES
11.5 Year	Closed	Paid

Window	First Day to Pay	Surcharge Starts	Last Day to Pay	Status	Fees
3.5 Year	01/15/2005	07/16/2005	01/17/2006	Closed	Paid
7.5 Year	01/15/2009	07/16/2009	01/15/2010	Closed	Paid
11.5 Year	01/15/2013	07/16/2013	01/15/2014	Closed	Paid

Patent Holder Information

Customer # 197

Entity Status UNDISCOUNTED

Phone Number 7037392234

Address CPA GLOBAL LIMITED
2318 Mill Road 12th Floor
ALEXANDRIA, VA 22314
UNITED STATES