

09/830114

patent application serial no.

*Department of Commerce
Patent and Trademark Office
for record*

04/26/2001 UEDUVIJE 00000045 09830114

01 FC:971	430.00 OP
02 FC:254	65.00 OP
03 FC:967	387.00 OP

ZTE Exhibit 1002

PATENT APPLICATION FEE DETERMINATION RECORD
Effective October 1, 2000

Application or Docket Number

09/830114

CLAIMS AS FILED - PART I

(Column 1) (Column 2)

TOTAL CLAIMS		
FOR	NUMBER FILED	NUMBER EXTRA
TOTAL CHARGEABLE CLAIMS	63 minus 20=	43
INDEPENDENT CLAIMS	3 minus 3=	
MULTIPLE DEPENDENT CLAIM PRESENT ☐		

* If the difference in column 1 is less than zero, enter "0" in column 2

SMALL ENTITY
TYPE ☐ OR

OTHER THAN
SMALL ENTITY

RATE	FEE
BASIC FEE	355.00
X\$ 9=	387
X40=	
+135=	
TOTAL	817

RATE	FEE
BASIC FEE	710.00
X\$18=	
X80=	
+270=	
TOTAL	

CLAIMS AS AMENDED - PART II

(Column 1) (Column 2) (Column 3)

AMENDMENT A	CLAIMS REMAINING AFTER AMENDMENT		HIGHEST NUMBER PREVIOUSLY PAID FOR	PRESENT EXTRA
	Total	*	Minus	**
	Independent	*	Minus	***
	FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM ☐			

SMALL ENTITY OR

OTHER THAN
SMALL ENTITY

RATE	ADDI-TIONAL FEE
X\$ 9=	
X40=	
+135=	
TOTAL	
ADDIT. FEE	

RATE	ADDI-TIONAL FEE
X\$18=	
X80=	
+270=	
TOTAL	
ADDIT. FEE	

(Column 1) (Column 2) (Column 3)

AMENDMENT B	CLAIMS REMAINING AFTER AMENDMENT		HIGHEST NUMBER PREVIOUSLY PAID FOR	PRESENT EXTRA
	Total	*	Minus	**
	Independent	*	Minus	***
	FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM ☐			

RATE	ADDI-TIONAL FEE
X\$ 9=	
X40=	
+135=	
TOTAL	
ADDIT. FEE	

RATE	ADDI-TIONAL FEE
X\$18=	
X80=	
+270=	
TOTAL	
ADDIT. FEE	

(Column 1) (Column 2) (Column 3)

AMENDMENT C	CLAIMS REMAINING AFTER AMENDMENT		HIGHEST NUMBER PREVIOUSLY PAID FOR	PRESENT EXTRA
	Total	*	Minus	**
	Independent	*	Minus	***
	FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM ☐			

RATE	ADDI-TIONAL FEE
X\$ 9=	
X40=	
+135=	
TOTAL	
ADDIT. FEE	

RATE	ADDI-TIONAL FEE
X\$18=	
X80=	
+270=	
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.

MULTIPLE DEPENDENT CLAIM FEE CALCULATION SHEET (FOR USE WITH FORM PTO-875)							SERIAL NO. 09/830114	FILING DATE		
							APPLICANT(S)			
CLAIMS										
	AS FILED		AFTER 1st AMENDMENT		AFTER 2nd AMENDMENT					
	IND.	DEP.	IND.	DEP.	IND.	DEP.		IND.	DEP.	
1	/						61	/		
2		/					52	/		
3		/					53	/		
4		/					54	/		
5		/					55	/		
6		/					56	/		
7		/					57	/		
8		/					58	/		
9		/					59	/		
10	/						60	/		
11		/					61	/		
12		/					62	/		
13		/					63	/		
14		/					64			
15		/					65			
16		/					66			
17		/					67			
18		/					68			
19		/					69			
20		/					70			
21		/					71			
22		/					72			
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32		/					82			
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36		/					86			
37		/					87			
38		/					88			
39		/					89			
40		/					90			
41		/					91			
42		/					92			
43		/					93			
44		/					94			
45		/					95			
46		/					96			
47		/					97			
48		/					98			
49		/					99			
50		/					100			
TOTAL IND.							TOTAL IND.	3		
TOTAL DEP.							TOTAL DEP.	66		
TOTAL CLAIMS							TOTAL CLAIMS	63		

PTO-1350 (3-78)

*MAY BE USED FOR ADDITIONAL CLAIMS OR AMENDMENTS

U.S. DEPARTMENT OF COMMERCE
Patent and Trademark Office

09/830114

Rec'd PCT/PTO 2 4 APR 2001

PATENT
4082-0129P

IN THE U.S. PATENT AND TRADEMARK OFFICE

#6/IDS
71

Applicant: Bruno BESSETTE et al.
Appl. No.: NEW Group:
Filed: April 24, 2001 Examiner:
For: A METHOD AND DEVICE FOR ADAPTIVE
BANDWIDTH PITCH SEARCH IN CODING
WIDEBAND SIGNALS

INFORMATION DISCLOSURE STATEMENT
(SUBMISSION CONCURRENT WITH THE
FILING OF A NEW PATENT APPLICATION)

Assistant Commissioner for Patents
Washington, DC 20231

April 24, 2001

Sir:

Pursuant to 37 C.F.R. §§ 1.97 and 1.98, applicant(s) hereby submit(s) an Information Disclosure Statement for consideration by the Examiner.

I. LIST OF PATENTS, PUBLICATIONS OR OTHER INFORMATION

The patents, publications, or other information submitted for consideration by the Office are listed on PTO-1449, attached hereto.

II. COPIES

- ☐ Submitted herewith is a legible copy of (i) each U.S. and foreign patent; (ii) each publication or that portion which caused it to be listed; and (iii) all other information or that portion which caused it to be listed.
- ☒ This application is a National Phase of a PCT application. Some or all of the documents listed on the PTO-1449 are not enclosed because they were cited in the International Search Report and copies should be forwarded from the International Search Authority. If copies are needed, please contact the undersigned.

III. CONCISE EXPLANATION OF THE RELEVANCE
(check at least one box)

a. ☒ **DOCUMENTS IN THE ENGLISH LANGUAGE**

The attached patents, publications, or other information in the English language do not require a statement of relevancy.

b. ☐ **DOCUMENTS NOT IN THE ENGLISH LANGUAGE**

A concise explanation of the relevance of all patents, publications, or other information listed that is not in the English language is as follows:

c. ☒ **ENGLISH LANGUAGE SEARCH REPORT**

An English language version of the search report or action that indicates the degree of relevance found by the foreign office is attached, thereby satisfying the requirement for a concise explanation. See MPEP 609(A)(3).

d. ☐ **OTHER**

The following additional information is provided for the Examiner's consideration.

FEES

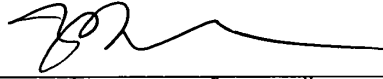
This Information Disclosure Statement is being filed concurrently with the filing of a new patent application; therefore, no fee is required.

If The Examiner has any questions concerning this IDS, he/she is requested to contact the undersigned. If it is determined that this IDS has been filed under the wrong rule, the PTO is requested to consider this IDS under the proper rule and charge the appropriate fee to Deposit Account No. 02-2448.

If necessary, the Commissioner is hereby authorized in this, concurrent, and future replies, to charge payment or credit any overpayment to Deposit Account No. 02-2448 for any additional fees required under 37 C.F.R. § 1.16 or under § 1.17; particularly, extension of time fees.

Respectfully submitted,

BIRCH, STEWART, KOLASCH & BIRCH, LLP

By 
F. Prince Butler, #25,666

April 24, 2001
4082-0129P

P.O. Box 747
Falls Church, VA 22040-0747
(703) 205-8000

Enclosures: ☒ Form PTO-1449(s)
☐ Documents
☒ Foreign Search Report
☐ Fee
☐ Other: _____

(Rev. 01/22/01)

[illegible]

PATENT COOPERATION TREATY

PCT**NOTIFICATION OF ELECTION**

(PCT Rule 61.2)

From the INTERNATIONAL BUREAU

To:

Assistant Commissioner for Patents
United States Patent and Trademark
Office
Box PCT
Washington, D.C.20231
ETATS-UNIS D'AMERIQUE

in its capacity as elected Office

Date of mailing (day/month/year)
02 June 2000 (02.06.00)

International application No.
PCT/CA99/01008

Applicant's or agent's file reference
GP/12916.10

International filing date (day/month/year)
27 October 1999 (27.10.99)

Priority date (day/month/year)
27 October 1998 (27.10.98)

Applicant

BESSETTE, Bruno et al

1. The designated Office is hereby notified of its election made:

☒ in the demand filed with the International Preliminary Examining Authority on:

28 April 2000 (28.04.00)

☐ in a notice effecting later election filed with the International Bureau on:
2. The election ☒ was
☐ was not

made before the expiration of 19 months from the priority date or, where Rule 32 applies, within the time limit under Rule 32.2(b).

The International Bureau of WIPO
34, chemin des Colombettes
1211 Geneva 20, Switzerland

Facsimile No.: (41-22) 740.14.35

Authorized officer

Beate Giffo-Schmitt

Telephone No.: (41-22) 338.83.38

PATENT COOPERATION TREATY

PCT

NOTIFICATION OF THE RECORDING OF A CHANGE

(PCT Rule 92bis.1 and
Administrative Instructions, Section 422)

From the INTERNATIONAL BUREAU

To:

DUBUC, Jean, H.
Goudreau Gage Dubuc
The Stock Exchange Tower
Suite 3400
800 Place Victoria
Montreal, Quebec H4Z 1E9
CANADA

Date of mailing (day/month/year) 02 June 2000 (02.06.00)	Applicant's or agent's file reference GP/12916.10
International application No. PCT/CA99/01008	IMPORTANT NOTIFICATION International filing date (day/month/year) 27 October 1999 (27.10.99)

1. The following indications appeared on record concerning:

☐ the applicant
 ☐ the inventor
 ☒ the agent
 ☐ the common representative

Name and Address

DUBUC, Jean, H.
Goudreau Gage Dubuc & Martineau
Walker
The Stock Exchange Tower
Suite 3400
800 Place Victoria
Montreal, Quebec H4Z 1E9
Canada

State of Nationality

State of Residence

Telephone No.

514 397 7609

Facsimile No.

514 397 4382

Teleprinter No.

2. The International Bureau hereby notifies the applicant that the following change has been recorded concerning:

☐ the person
 ☐ the name
 ☒ the address
 ☐ the nationality
 ☐ the residence

Name and Address

DUBUC, Jean, H.
Goudreau Gage Dubuc
The Stock Exchange Tower
Suite 3400
800 Place Victoria
Montreal, Quebec H4Z 1E9
Canada

State of Nationality

State of Residence

Telephone No.

514 397 7609

Facsimile No.

514 397 4382

Teleprinter No.

3. Further observations, if necessary:

The change in the address of the agent in the demand form has been considered as a request for change under Rule 92.bis. In case of disagreement, the International Bureau should be immediately notified accordingly.

4. A copy of this notification has been sent to:

☒ the receiving Office
 ☐ the designated Offices concerned
☐ the International Searching Authority
 ☒ the elected Offices concerned
☒ the International Preliminary Examining Authority
 ☐ other:

The International Bureau of WIPO
34, chemin des Colombettes
1211 Geneva 20, Switzerland

Facsimile No.: (41-22) 740.14.35

Authorized officer

Beate Giffo-Schmitt

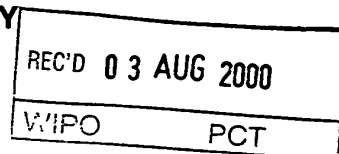
Telephone No.: (41-22) 338.83.38

PATENT COOPERATION TREATY

PCT

INTERNATIONAL PRELIMINARY EXAMINATION REPORT

(PCT Article 36 and Rule 70)



15

Applicant's or agent's file reference GP/12916.10	FOR FURTHER ACTION		See Notification of Transmittal of International Preliminary Examination Report (Form PCT/IPEA/416)
International application No. PCT/CA99/01008	International filing date (day/month/year) 27/10/1999	Priority date (day/month/year) 27/10/1998	
International Patent Classification (IPC) or national classification and IPC G10L11/04			
Applicant VOICEAGE CORPORATION et al.			

1. This international preliminary examination report has been prepared by this International Preliminary Examining Authority and is transmitted to the applicant according to Article 36.

2. This REPORT consists of a total of 5 sheets, including this cover sheet.

- ☐ This report is also accompanied by ANNEXES, i.e. sheets of the description, claims and/or drawings which have been amended and are the basis for this report and/or sheets containing rectifications made before this Authority (see Rule 70.16 and Section 607 of the Administrative Instructions under the PCT).

These annexes consist of a total of sheets.

3. This report contains indications relating to the following items:

- I ☒ Basis of the report
- II ☐ Priority
- III ☐ Non-establishment of opinion with regard to novelty, inventive step and industrial applicability
- IV ☐ Lack of unity of invention
- V ☒ Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement
- VI ☐ Certain documents cited
- VII ☒ Certain defects in the international application
- VIII ☒ Certain observations on the international application

Date of submission of the demand 28/04/2000	Date of completion of this report 01.08.2000
Name and mailing address of the international preliminary examining authority: European Patent Office D-80298 Munich Tel. +49 89 2399 - 0 Tx: 523656 epmu d Fax: +49 89 2399 - 4465	Authorized officer Greiser, N Telephone No. +49 89 2399 7402



**INTERNATIONAL PRELIMINARY
EXAMINATION REPORT**

International application No. PCT/CA99/01008

I. Basis of the report

1. This report has been drawn on the basis of (*substitute sheets which have been furnished to the receiving Office in response to an invitation under Article 14 are referred to in this report as "originally filed" and are not annexed to the report since they do not contain amendments.*):

Description, pages:

1-43 as originally filed

Claims, No.:

1-63 as originally filed

Drawings, sheets:

1/4-4/4 as originally filed

2. The amendments have resulted in the cancellation of:

- ☐ the description, pages:
☐ the claims, Nos.:
☐ the drawings, sheets:

3. ☐ This report has been established as if (some of) the amendments had not been made, since they have been considered to go beyond the disclosure as filed (Rule 70.2(c)):

4. Additional observations, if necessary:

**INTERNATIONAL PRELIMINARY
EXAMINATION REPORT**

International application No. PCT/CA99/01008

V. Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement

1. Statement

Novelty (N)	Yes: Claims 1-63
	No: Claims
Inventive step (IS)	Yes: Claims 1-63
	No: Claims
Industrial applicability (IA)	Yes: Claims 1-63
	No: Claims

2. Citations and explanations

see separate sheet

VII. Certain defects in the international application

The following defects in the form or contents of the international application have been noted:

see separate sheet

VIII. Certain observations on the international application

The following observations on the clarity of the claims, description, and drawings or on the question whether the claims are fully supported by the description, are made:

see separate sheet

**INTERNATIONAL PRELIMINARY
EXAMINATION REPORT - SEPARATE SHEET**

International application No. PCT/CA99/01008

To Section V:

1. The technical filed of invention is a pitch analysis device for analysing a speech signal.

In document D1= EP-A-0421444, a pitch analysis technique for analysing a speech signal is disclosed. The inventors realize that the prior-art pitch analysis technique is not efficient for wideband speech signals since the pitch contents in the spectrum of voice segments in wideband speech signals do not extend over a whole spectrum range, i.e. a harmonic structure in the wideband speech signal exists only up to a certain frequency. To solve the technical problem, the inventors suggest to filter respective sets of pitch codebook parameters in at least two signal paths, calculate the corresponding pitch prediction errors and choose the signal path with the lowest calculated pitch prediction error for selecting the associated set of pitch parameters.

Under the interpretation of Section VIII of this Report, the subject-matter of claim 1 is novel. Moreover, the subject-matter of claim 1 is non-obvious and, hence, inventive.

2. Claim 10 claims a pitch analysis method for producing an optimal set of pitch codebook parameters which corresponds to apparatus claim 1. Under the interpretation of Section VIII of this Report, claim 10 is novel and inventive.
3. Dependent claims 2-9, 11-63 are novel and appear to involve an inventive step.

To Section VII:

4. To increase the intelligibility of the claims reference signs in parentheses should be inserted (Rule 6.2 (b) PCT).

To Section VIII:

5. The subject-matter of claim 1 is too broad and, hence, not clear.

**INTERNATIONAL PRELIMINARY
EXAMINATION REPORT - SEPARATE SHEET**

International application No. PCT/CA99/01008

In its present form, claim 1 claims a pitch analysis device. Since the description specifies a pitch analysis technique for analysing speech signals claim 1 is not sufficiently supported by the description. To remedy the deficiency, it is suggested to claim a pitch analysis device for analysing a speech signal to produce an optimal set of pitch codebook parameters (amendments are underlined). The same objection applies to method claim 10.

PATENT COOPERATION TREATY

PCT

INTERNATIONAL SEARCH REPORT

(PCT Article 18 and Rules 43 and 44)

Applicant's or agent's file reference GP/12916.10	FOR FURTHER ACTION see Notification of Transmittal of International Search Report (Form PCT/ISA/220) as well as, where applicable, item 5 below.	
International application No. PCT/CA 99/ 01008	International filing date (day/month/year) 27/10/1999	(Earliest) Priority Date (day/month/year) 27/10/1998
Applicant VOICEAGE CORPORATION et al.		

This International Search Report has been prepared by this International Searching Authority and is transmitted to the applicant according to Article 18. A copy is being transmitted to the International Bureau.

This International Search Report consists of a total of 2 sheets.



It is also accompanied by a copy of each prior art document cited in this report.

1. Basis of the report

- a. With regard to the **language**, the international search was carried out on the basis of the international application in the language in which it was filed, unless otherwise indicated under this item.



the international search was carried out on the basis of a translation of the international application furnished to this Authority (Rule 23.1(b)).

- b. With regard to any **nucleotide and/or amino acid sequence** disclosed in the international application, the international search was carried out on the basis of the sequence listing :



contained in the international application in written form.



filed together with the international application in computer readable form.



furnished subsequently to this Authority in written form.



furnished subsequently to this Authority in computer readable form.



the statement that the subsequently furnished written sequence listing does not go beyond the disclosure in the international application as filed has been furnished.



the statement that the information recorded in computer readable form is identical to the written sequence listing has been furnished

2. ☐ **Certain claims were found unsearchable** (See Box I).

3. ☐ **Unity of invention is lacking** (see Box II).

4. With regard to the **title**,



the text is approved as submitted by the applicant.



the text has been established by this Authority to read as follows:

5. With regard to the **abstract**,



the text is approved as submitted by the applicant.



the text has been established, according to Rule 38.2(b), by this Authority as it appears in Box III. The applicant may, within one month from the date of mailing of this international search report, submit comments to this Authority.

6. The figure of the **drawings** to be published with the abstract is Figure No.



as suggested by the applicant.



because the applicant failed to suggest a figure.



because this figure better characterizes the invention.

3



None of the figures.

INTERNATIONAL SEARCH REPORT

International Application No

PCT/CA 99/01008

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 G10L11/04

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC 7 G10L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	EP 0 331 858 A (IBM) 13 September 1989 (1989-09-13) abstract; claims 5-7 page 3, line 16 - line 25 page 9, line 35 - line 43 ---	1,10
A	EP 0 722 165 A (DIGITAL VOICE SYSTEMS INC) 17 July 1996 (1996-07-17) abstract ---	1,10
A	EP 0 421 444 A (FUJITSU LTD) 10 April 1991 (1991-04-10) abstract; claim 1 -----	1,10

☐ Further documents are listed in the continuation of box C.

☒ Patent family members are listed in annex.

* Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document but published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

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

"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.

"&" document member of the same patent family

Date of the actual completion of the international search

31 January 2000

Date of mailing of the international search report

07/02/2000

Name and mailing address of the ISA

European Patent Office, P.B. 5818 Patentlaan 2
NL - 2280 HV Rijswijk
Tel. (+31-70) 340-2040, Tx. 31 651 epo nl,
Fax: (+31-70) 340-3016

Authorized officer

Van Doremalen, J

INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No

PCT/CA 99/01008

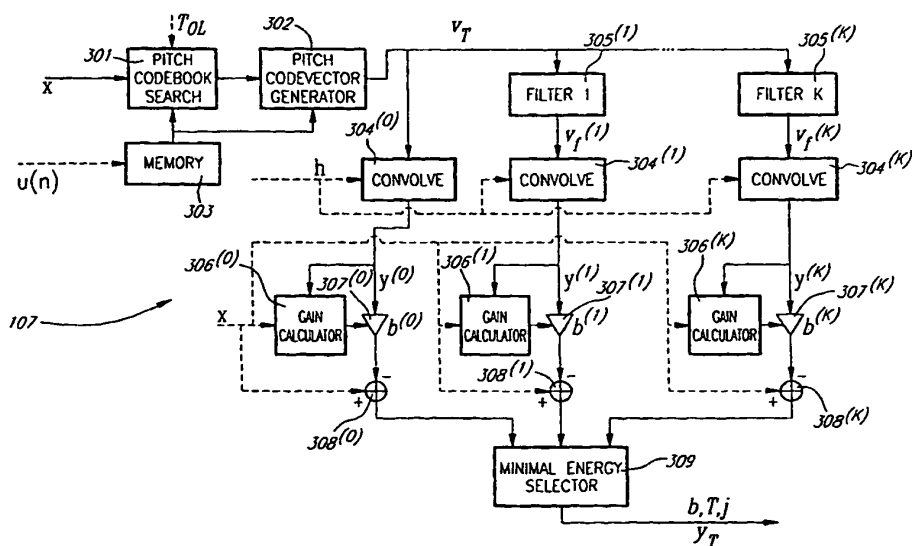
Patent document cited in search report		Publication date	Patent family member(s)	Publication date
EP 0331858	A	13-09-1989	DE 3883519 D	30-09-1993
			DE 3883519 T	17-03-1994
			JP 1233500 A	19-09-1989
			JP 2111572 C	21-11-1996
			JP 8033759 B	29-03-1996
			US 4965789 A	23-10-1990
EP 0722165	A	17-07-1996	AU 696092 B	03-09-1998
			AU 4085396 A	18-07-1996
			CA 2167025 A	13-07-1996
			US 5826222 A	20-10-1998
EP 0421444	A	10-04-1991	JP 3123113 A	24-05-1991
			CA 2026823 A, C	06-04-1991
			US 5231692 A	27-07-1993



INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(51) International Patent Classification ⁷ : G10L 11/04		A1	(11) International Publication Number: WO 00/25298
			(43) International Publication Date: 4 May 2000 (04.05.00)
(21) International Application Number: PCT/CA99/01008 (22) International Filing Date: 27 October 1999 (27.10.99) (30) Priority Data: 2,252,170 27 October 1998 (27.10.98) CA (71) Applicant (for all designated States except US): VOICEAGE CORPORATION [CA/CA]; Suite 200, 750, Chemin Lucerne, Ville Mont-Royal, Quebec H3R 2H6 (CA). (72) Inventors; and (75) Inventors/Applicants (for US only): BESSETTE, Bruno [CA/CA]; 1546 Pérodeau, Rock Forest, Quebec J1N 1L2 (CA). SALAMI, Redwan [CA/CA]; 963, Léo Laliberté, Sherbrooke, Quebec J1J 4L3 (CA). LEFEBVRE, Roch [CA/CA]; 259, avenue de la Bourgade, Canton de Magog, Quebec J1K 5R9 (CA). (74) Agents: DUBUC, Jean, H. et al.; Goudreau Gage Dubuc & Martineau Walker, The Stock Exchange Tower, Suite 3400, 800 Place Victoria, Montreal, Quebec H4Z 1E9 (CA).		(81) Designated States: AE, AL, AM, AT, AU, AZ, BA, BB, BG, BR, BY, CA, CH, CN, CR, CU, CZ, DE, DK, DM, EE, ES, FI, GB, GD, GE, GH, GM, HR, HU, ID, IL, IN, IS, JP, KE, KG, KP, KR, KZ, LC, LK, LR, LS, LT, LU, LV, MA, MD, MG, MK, MN, MW, MX, NO, NZ, PL, PT, RO, RU, SD, SE, SG, SI, SK, SL, TJ, TM, TR, TT, TZ, UA, UG, US, UZ, VN, YU, ZA, ZW, ARIPO patent (GH, GM, KE, LS, MW, SD, SL, SZ, TZ, UG, ZW), Eurasian patent (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European patent (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, GW, ML, MR, NE, SN, TD, TG). Published With international search report.	

(54) Title: A METHOD AND DEVICE FOR ADAPTIVE BANDWIDTH PITCH SEARCH IN CODING WIDEBAND SIGNALS



(57) Abstract

An improved pitch search method and device for digitally encoding a wideband signal, in particular but not exclusively a speech signal, in view of transmitting, or storing, and synthesizing this wideband sound signal. The new method and device which achieve efficient modeling of the harmonic structure of the speech spectrum uses several forms of low pass filters applied to a pitch codevector, the one yielding higher prediction gain (i.e. the lowest pitch prediction error) is selected and the associated pitch codebook parameters are forwarded.

FOR THE PURPOSES OF INFORMATION ONLY

Codes used to identify States party to the PCT on the front pages of pamphlets publishing international applications under the PCT.

AL	Albania	ES	Spain	LS	Lesotho	SI	Slovenia
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AT	Austria	FR	France	LU	Luxembourg	SN	Senegal
AU	Australia	GA	Gabon	LV	Latvia	SZ	Swaziland
AZ	Azerbaijan	GB	United Kingdom	MC	Monaco	TD	Chad
BA	Bosnia and Herzegovina	GE	Georgia	MD	Republic of Moldova	TG	Togo
BB	Barbados	GH	Ghana	MG	Madagascar	TJ	Tajikistan
BE	Belgium	GN	Guinea	MK	The former Yugoslav Republic of Macedonia	TM	Turkmenistan
BF	Burkina Faso	GR	Greece	ML	Mali	TR	Turkey
BG	Bulgaria	HU	Hungary	MN	Mongolia	TT	Trinidad and Tobago
BJ	Benin	IE	Ireland	MR	Mauritania	UA	Ukraine
BR	Brazil	IL	Israel	MW	Malawi	UG	Uganda
BY	Belarus	IS	Iceland	MX	Mexico	US	United States of America
CA	Canada	IT	Italy	NE	Niger	UZ	Uzbekistan
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CG	Congo	KE	Kenya	NO	Norway	YU	Yugoslavia
CH	Switzerland	KG	Kyrgyzstan	NZ	New Zealand	ZW	Zimbabwe
CI	Côte d'Ivoire	KP	Democratic People's Republic of Korea	PL	Poland		
CM	Cameroon	KR	Republic of Korea	PT	Portugal		
CN	China	KZ	Kazakistan	RO	Romania		
CU	Cuba	LC	Saint Lucia	RU	Russian Federation		
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DE	Germany	LK	Sri Lanka	SE	Sweden		
DK	Denmark	LR	Liberia	SG	Singapore		
EE	Estonia						

A METHOD AND DEVICE FOR ADAPTIVE BANDWIDTH
PITCH SEARCH IN CODING WIDEBAND SIGNALS

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BACKGROUND OF THE INVENTION

1. Field of the invention:

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The present invention relates to an efficient technique for digitally encoding a wideband signal, in particular but not exclusively a speech signal, in view of transmitting, or storing, and synthesizing this wideband sound signal. More specifically, this invention deals with an improved pitch search device and method.

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2. Brief description of the prior art:

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The demand for efficient digital wideband speech/audio encoding techniques with a good subjective quality/bit rate trade-off is increasing for numerous applications such as audio/video teleconferencing, multimedia, and wireless applications, as well as Internet and packet network applications. Until recently, telephone

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bandwidths filtered in the range 200-3400 Hz were mainly used in speech coding applications. However, there is an increasing demand for wideband speech applications in order to increase the intelligibility and

naturalness of the speech signals. A bandwidth in the range 50-7000 Hz was found sufficient for delivering a face-to-face speech quality. For audio signals, this range gives an acceptable audio quality, but still lower than the CD quality which operates on the range 20-20000 Hz.

5 A speech encoder converts a speech signal into a digital bitstream which is transmitted over a communication channel (or stored in a storage medium). The speech signal is digitized (sampled and quantized with usually 16-bits per sample) and the speech encoder has the role of representing these digital samples with a smaller number of
10 bits while maintaining a good subjective speech quality. The speech decoder or synthesizer operates on the transmitted or stored bit stream and converts it back to a sound signal.

 One of the best prior art techniques capable of achieving a
15 good quality/bit rate trade-off is the so-called Code Excited Linear Prediction (CELP) technique. According to this technique, the sampled speech signal is processed in successive blocks of L samples usually called *frames* where L is some predetermined number (corresponding to 10-30 ms of speech). In CELP, a linear prediction (LP) filter is computed
20 and transmitted every frame. The L -sample frame is then divided into smaller blocks called *subframes* of size N samples, where $L=kN$ and k is the number of subframes in a frame (N usually corresponds to 4-10 ms of speech). An excitation signal is determined in each subframe, which usually consists of two components: one from the past excitation (also
25 called pitch contribution or adaptive codebook) and the other from an innovation codebook (also called fixed codebook). This excitation signal

is transmitted and used at the decoder as the input of the LP synthesis filter in order to obtain the synthesized speech.

An innovation codebook in the CELP context, is an indexed set of N -sample-long sequences which will be referred to as N -dimensional codevectors. Each codebook sequence is indexed by an integer k ranging from 1 to M where M represents the size of the codebook often expressed as a number of bits b , where $M=2^b$.

To synthesize speech according to the CELP technique, each block of N samples is synthesized by filtering an appropriate codevector from a codebook through time varying filters modeling the spectral characteristics of the speech signal. At the encoder end, the synthetic output is computed for all, or a subset, of the codevectors from the codebook (codebook search). The retained codevector is the one producing the synthetic output closest to the original speech signal according to a perceptually weighted distortion measure. This perceptual weighting is performed using a so-called perceptual weighting filter, which is usually derived from the LP filter.

The CELP model has been very successful in encoding telephone band sound signals, and several CELP-based standards exist in a wide range of applications, especially in digital cellular applications. In the telephone band, the sound signal is band-limited to 200-3400 Hz and sampled at 8000 samples/sec. In wideband speech/audio applications, the sound signal is band-limited to 50-7000 Hz and sampled at 16000 samples/sec.

Some difficulties arise when applying the telephone-band optimized CELP model to wideband signals, and additional features need to be added to the model in order to obtain high quality wideband signals. Wideband signals exhibit a much wider dynamic range compared to telephone-band signals, which results in precision problems when a fixed-point implementation of the algorithm is required (which is essential in wireless applications). Further, the CELP model will often spend most of its encoding bits on the low-frequency region, which usually has higher energy contents, resulting in a low-pass output signal. To overcome this problem, the perceptual weighting filter has to be modified in order to suit wideband signals, and pre-emphasis techniques which boost the high frequency regions become important to reduce the dynamic range, yielding a simpler fixed-point implementation, and to ensure a better encoding of the higher frequency contents of the signal. Further, the pitch contents in the spectrum of voiced segments in wideband signals do not extend over the whole spectrum range, and the amount of voicing shows more variation compared to narrow-band signals. Therefore, in case of wideband signals, existing pitch search structures are not very efficient. Thus, it is important to improve the closed-loop pitch analysis to better accommodate the variations in the voicing level.

OBJECTS OF THE INVENTION

An object of the present invention is therefore to provide a method and device for efficiently encoding wideband (7000 Hz) sound signals using

CELP-type encoding techniques, using improved pitch analysis in order to obtain high a quality reconstructed sound signal.

SUMMARY OF THE INVENTION

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More specifically, in accordance with the present invention, there is provided a method for selecting an optimal set of pitch codebook parameters associated to a signal path, from at least two signal paths, having the lowest calculated pitch prediction error. The pitch prediction error is calculated in response to a pitch codevector from a pitch codebook search device. In at least one of the two signal paths, the pitch prediction error is filtered before supplying the pitch codevector for calculation of said pitch prediction error of said one path. Finally, the pitch prediction errors calculated in said at least two signal paths are compared, the signal path having the lowest calculated pitch prediction error is chosen, and the set of pitch codebook parameters associated to the choosen signal path are selected.

The pitch analysis device of the invention, for producing an optimal set of pitch codebook parameters, comprises:

a) at least two signal paths associated to respective sets of pitch codebook parameters, wherein:

- i) each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device; and
- ii) at least one of the two paths comprises a filter for filtering the pitch codevector before supplying the pitch

codevector to the path's pitch prediction error calculating device; and

- b) a selector for comparing the pitch prediction errors calculated in the signal paths, for choosing the signal path having the lowest calculated pitch prediction error, and for selecting the set of pitch codebook parameters associated to the chosen signal path.

The new method and device which achieve efficient modeling of the harmonic structure of the speech spectrum uses several forms of low pass filters applied to the past excitation and the one yielding higher prediction gain is selected. When subsample pitch resolution is used, the low pass filters can be incorporated into the interpolation filters used to obtain the higher pitch resolution.

In a preferred embodiment of the invention, each pitch prediction error calculating device of the pitch analysis device described above comprises:

- a) a convolution unit for convolving the pitch codevector with a weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;
- b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;
- c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector;
- and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

In another preferred embodiment of the invention, the pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^T y^{(j)} / \|y^{(j)}\|^2 \quad \text{where } j = 0, 1, 2, \dots, K, \text{ and } K \text{ corresponds to a number of signal paths,}$$

and where x is said pitch search target vector, and $y^{(j)}$ is said convolved pitch codevector.

The present invention further relates to an encoder, having the pitch analysis device described above, for encoding a wideband input signal and comprising:

- a) a linear prediction synthesis filter calculator responsive to the wideband signal for producing linear prediction synthesis filter coefficients;
- b) a perceptual weighting filter, responsive to the wideband signal and the linear prediction synthesis filter coefficients, for producing a perceptually weighted signal;
- c) an impulse response generator responsive to the linear prediction synthesis filter coefficients for producing a weighted synthesis filter impulse response signal;
- d) a pitch search unit for producing pitch codebook parameters, comprising:

i) a pitch codebook search device responsive to the perceptually weighted signal and the linear prediction synthesis filter coefficients for producing the pitch codevector and an innovative search target vector; and

ii) the pitch analysis device responsive to the pitch codevector for selecting, from the sets of pitch codebook parameters, the set of pitch codebook parameters associated to the path having the lowest calculated pitch prediction error;

d) an innovative codebook search device, responsive to the weighted synthesis filter impulse response signal, and the innovative search target vector, for producing innovative codebook parameters; and

e) a signal forming device for producing an encoded wideband signal comprising the set of pitch codebook parameters associated to the path having the lowest pitch prediction error, the innovative codebook parameters, and the linear prediction synthesis filter coefficients.

The present invention still further relates to a cellular communication system, a cellular mobile transmitter/receiver unit, a cellular network element, and a bidirectional wireless communication sub-system comprising the above described decoder.

The objects, advantages and other features of the present invention will become more apparent upon reading of the following non restrictive description of a preferred embodiment thereof, given by way of example only with reference to the accompanying drawings.

BRIEF DESCRIPTION OF THE DRAWINGS

In the appended drawings:

5 Figure 1 is a schematic block diagram of a preferred embodiment of wideband encoding device;

 Figure 2 is a schematic block diagram of a preferred embodiment of wideband decoding device;

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 Figure 3 is a schematic block diagram of a preferred embodiment of pitch analysis device; and

 Figure 4 is a simplified, schematic block diagram of a cellular communication system in which the wideband encoding device of Figure 1 and the wideband decoding device of Figure 2 can be used.

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20 DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

 As well known to those of ordinary skill in the art, a cellular communication system such as 401 (see Figure 4) provides a telecommunication service over a large geographic area by dividing that large geographic area into a number C of smaller cells. The C smaller cells

25

are serviced by respective cellular base stations 402_1 , 402_2 ... 402_c to provide each cell with radio signalling, audio and data channels.

Radio signalling channels are used to page mobile radiotelephones (mobile transmitter/receiver units) such as 403 within the limits of the coverage area (cell) of the cellular base station 402, and to place calls to other radiotelephones 403 located either inside or outside the base station's cell or to another network such as the Public Switched Telephone Network (PSTN) 404.

Once a radiotelephone 403 has successfully placed or received a call, an audio or data channel is established between this radiotelephone 403 and the cellular base station 402 corresponding to the cell in which the radiotelephone 403 is situated, and communication between the base station 402 and radiotelephone 403 is conducted over that audio or data channel. The radiotelephone 403 may also receive control or timing information over a signalling channel while a call is in progress.

If a radiotelephone 403 leaves a cell and enters another adjacent cell while a call is in progress, the radiotelephone 403 hands over the call to an available audio or data channel of the new cell base station 402. If a radiotelephone 403 leaves a cell and enters another adjacent cell while no call is in progress, the radiotelephone 403 sends a control message over the signalling channel to log into the base station 402 of the new cell. In this manner mobile communication over a wide geographical area is possible.

The cellular communication system 401 further comprises a control terminal 405 to control communication between the cellular base stations

402 and the PSTN 404, for example during a communication between a radiotelephone 403 and the PSTN 404, or between a radiotelephone 403 located in a first cell and a radiotelephone 403 situated in a second cell.

Of course, a bidirectional wireless radio communication subsystem is required to establish an audio or data channel between a base station 402 of one cell and a radiotelephone 403 located in that cell. As illustrated in very simplified form in Figure 4, such a bidirectional wireless radio communication subsystem typically comprises in the radiotelephone 403:

- a transmitter 406 including:

- an encoder 407 for encoding the voice signal; and
- a transmission circuit 408 for transmitting the encoded voice signal from the encoder 407 through an antenna such as 409; and

- a receiver 410 including:

- a receiving circuit 411 for receiving a transmitted encoded voice signal usually through the same antenna 409; and
- a decoder 412 for decoding the received encoded voice signal from the receiving circuit 411.

The radiotelephone further comprises other conventional radiotelephone circuits 413 to which the encoder 407 and decoder 412 are connected and for processing signals therefrom, which circuits 413 are well known to those of ordinary skill in the art and, accordingly, will not be further described in the present specification.

Also, such a bidirectional wireless radio communication subsystem typically comprises in the base station 402:

- a transmitter 414 including:
 - an encoder 415 for encoding the voice signal; and
 - a transmission circuit 416 for transmitting the encoded voice signal from the encoder 415 through an antenna such as 417; and
- 5 - a receiver 418 including:
 - a receiving circuit 419 for receiving a transmitted encoded voice signal through the same antenna 417 or through another antenna (not shown); and
 - a decoder 420 for decoding the received encoded voice signal
- 10 from the receiving circuit 419.

The base station 402 further comprises, typically, a base station controller 421, along with its associated database 422, for controlling communication between the control terminal 405 and the transmitter 414 and
15 receiver 418.

As well known to those of ordinary skill in the art, voice encoding is required in order to reduce the bandwidth necessary to transmit sound signal, for example voice signal such as speech, across the bidirectional
20 wireless radio communication subsystem, i.e., between a radiotelephone 403 and a base station 402.

LP voice encoders (such as 415 and 407) typically operating at 13 kbits/second and below such as Code-Excited Linear Prediction (CELP)
25 encoders typically use a LP synthesis filter to model the short-term spectral envelope of the voice signal. The LP information is transmitted, typically,

every 10 or 20 ms to the decoder (such 420 and 412) and is extracted at the decoder end.

The novel techniques disclosed in the present specification may apply to different LP-based coding systems. However, a CELP-type coding system is used in the preferred embodiment for the purpose of presenting a non-limitative illustration of these techniques. In the same manner, such techniques can be used with sound signals other than voice and speech as well with other types of wideband signals.

Figure 1 shows a general block diagram of a CELP-type speech encoding device 100 modified to better accommodate wideband signals.

The sampled input speech signal 114 is divided into successive L -sample blocks called "frames". In each frame, different parameters representing the speech signal in the frame are computed, encoded, and transmitted. LP parameters representing the LP synthesis filter are usually computed once every frame. The frame is further divided into smaller blocks of N samples (blocks of length N), in which excitation parameters (pitch and innovation) are determined. In the CELP literature, these blocks of length N are called "subframes" and the N -sample signals in the subframes are referred to as N -dimensional vectors. In this preferred embodiment, the length N corresponds to 5 ms while the length L corresponds to 20 ms, which means that a frame contains four subframes ($N=80$ at the sampling rate of 16 kHz and 64 after down-sampling to 12.8 kHz). Various N -dimensional vectors occur in the encoding procedure. A list of the vectors which appear in Figures 1 and 2 as well as a list of transmitted parameters are given herein below:

List of the main N -dimensional vectors

- s** Wideband signal input speech vector (after down-sampling, pre-processing, and preemphasis);
- 5 **s_w** Weighted speech vector;
- s_o** Zero-input response of weighted synthesis filter;
- s_p** Down-sampled pre-processed signal;
- Oversampled synthesized speech signal;
- 10 **s'** Synthesis signal before deemphasis;
- s_d** Deemphasized synthesis signal;
- s_h** Synthesis signal after deemphasis and postprocessing;
- x** Target vector for pitch search;
- x'** Target vector for innovation search;
- 15 **h** Weighted synthesis filter impulse response;
- v_T** Adaptive (pitch) codebook vector at delay T ;
- y_T** Filtered pitch codebook vector (v_T convolved with h);
- c_k** Innovative codevector at index k (k -th entry from the innovation codebook);
- 20 **c_f** Enhanced scaled innovation codevector;
- u** Excitation signal (scaled innovation and pitch codevectors);
- u'** Enhanced excitation;
- z** Band-pass noise sequence;
- w'** White noise sequence; and
- 25 **w** Scaled noise sequence.

List of transmitted parameters

- STP Short term prediction parameters (defining $A(z)$);
 T Pitch lag (or pitch codebook index);
 b Pitch gain (or pitch codebook gain);
5 j Index of the low-pass filter used on the pitch codevector;
 k Codevector index (innovation codebook entry); and
 g Innovation codebook gain.

10 In this preferred embodiment, the STP parameters are transmitted once per frame and the rest of the parameters are transmitted four times per frame (every subframe).

ENCODER SIDE

15 The sampled speech signal is encoded on a block by block basis by the encoding device 100 of Figure 1 which is broken down into eleven modules numbered from 101 to 111.

20 The input speech is processed into the above mentioned L -sample blocks called frames.

Referring to Figure 1, the sampled input speech signal 114 is down-sampled in a down-sampling module 101. For example, the signal is down-sampled from 16 kHz down to 12.8 kHz, using techniques well known to
25 those of ordinary skill in the art. Down-sampling down to another frequency can of course be envisaged. Down-sampling increases the coding efficiency, since a smaller frequency bandwidth is encoded. This also

reduces the algorithmic complexity since the number of samples in a frame is decreased. The use of down-sampling becomes significant when the bit rate is reduced below 16 kbit/s, although down-sampling is not essential above 16 kbit/s.

- 5 After down-sampling, the 320-sample frame of 20 ms is reduced to 256-sample frame (down-sampling ratio of 4/5).

 The input frame is then supplied to the optional pre-processing block 102. Pre-processing block 102 may consist of a high-pass filter with a 50 Hz
10 cut-off frequency. High-pass filter 102 removes the unwanted sound components below 50 Hz.

 The down-sampled pre-processed signal is denoted by $s_p(n)$, $n=0, 1, 2, \dots, L-1$, where L is the length of the frame (256 at a sampling frequency of
15 12.8 kHz). In a preferred embodiment of the preemphasis filter 103, the signal $s_p(n)$ is preemphasized using a filter having the following transfer function:

20
$$P(z) = 1 - \mu z^{-1}$$

 where μ is a preemphasis factor with a value located between 0 and 1 (a
25 typical value is $\mu = 0.7$). A higher-order filter could also be used. It should be pointed out that high-pass filter 102 and preemphasis filter 103 can be interchanged to obtain more efficient fixed-point implementations.

The function of the preemphasis filter 103 is to enhance the high frequency contents of the input signal. It also reduces the dynamic range of the input speech signal, which renders it more suitable for fixed-point implementation. Without preemphasis, LP analysis in fixed-point using single-precision arithmetic is difficult to implement.

5

Preemphasis also plays an important role in achieving a proper overall perceptual weighting of the quantization error, which contributes to improved sound quality. This will be explained in more detail herein below.

10

The output of the preemphasis filter 103 is denoted $s(n)$. This signal is used for performing LP analysis in calculator module 104. LP analysis is a technique well known to those of ordinary skill in the art. In this preferred embodiment, the autocorrelation approach is used. In the autocorrelation approach, the signal $s(n)$ is first windowed using a Hamming window (having usually a length of the order of 30-40 ms). The autocorrelations are computed from the windowed signal, and Levinson-Durbin recursion is used to compute LP filter coefficients, a_i , where $i=1, \dots, p$, and where p is the LP order, which is typically 16 in wideband coding. The parameters a_i are the coefficients of the transfer function of the LP filter, which is given by the following relation:

15

20

$$A(z) = 1 + \sum_{i=1}^p a_i z^{-i}$$

25

LP analysis is performed in calculator module 104, which also performs the quantization and interpolation of the LP filter coefficients. The LP filter coefficients are first transformed into another equivalent domain more suitable for quantization and interpolation purposes. The line spectral pair (LSP) and immittance spectral pair (ISP) domains are two domains in which quantization and interpolation can be efficiently performed. The 16 LP filter coefficients, a_n , can be quantized in the order of 30 to 50 bits using split or multi-stage quantization, or a combination thereof. The purpose of the interpolation is to enable updating the LP filter coefficients every subframe while transmitting them once every frame, which improves the encoder performance without increasing the bit rate. Quantization and interpolation of the LP filter coefficients is believed to be otherwise well known to those of ordinary skill in the art and, accordingly, will not be further described in the present specification.

The following paragraphs will describe the rest of the coding operations performed on a subframe basis. In the following description, the filter $A(z)$ denotes the unquantized interpolated LP filter of the subframe, and the filter $\hat{A}(z)$ denotes the quantized interpolated LP filter of the subframe.

20 Perceptual Weighting:

In analysis-by-synthesis encoders, the optimum pitch and innovation parameters are searched by minimizing the mean squared error between the input speech and synthesized speech in a perceptually weighted domain. This is equivalent to minimizing the error between the weighted input speech and weighted synthesis speech.

The weighted signal $s_w(n)$ is computed in a perceptual weighting filter 105. Traditionally, the weighted signal $s_w(n)$ is computed by a weighting filter having a transfer function $W(z)$ in the form:

$$W(z) = A(z/\gamma_1) / A(z/\gamma_2) \quad \text{where} \quad 0 < \gamma_2 < \gamma_1 \leq 1$$

As well known to those of ordinary skill in the art, in prior art analysis-by-synthesis (AbS) encoders, analysis shows that the quantization error is weighted by a transfer function $W^{-1}(z)$, which is the inverse of the transfer function of the perceptual weighting filter 105. This result is well described by B.S. Atal and M.R. Schroeder in "Predictive coding of speech and subjective error criteria", IEEE Transaction ASSP, vol. 27, no. 3, pp. 247-254, June 1979. Transfer function $W^{-1}(z)$ exhibits some of the formant structure of the input speech signal. Thus, the masking property of the human ear is exploited by shaping the quantization error so that it has more energy in the formant regions where it will be masked by the strong signal energy present in these regions. The amount of weighting is controlled by the factors γ_1 and γ_2 .

20

The above traditional perceptual weighting filter 105 works well with telephone band signals. However, it was found that this traditional perceptual weighting filter 105 is not suitable for efficient perceptual weighting of wideband signals. It was also found that the traditional perceptual weighting filter 105 has inherent limitations in modelling the formant structure and the required spectral tilt concurrently. The spectral tilt is more pronounced in wideband signals due to the wide dynamic range

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between low and high frequencies. The prior art has suggested to add a tilt filter into $W(z)$ in order to control the tilt and formant weighting of the wideband input signal separately.

5 A novel solution to this problem is, in accordance with the present invention, to introduce the preemphasis filter 103 at the input, compute the LP filter $A(z)$ based on the preemphasized speech $s(n)$, and use a modified filter $W(z)$ by fixing its denominator.

10 LP analysis is performed in module 104 on the preemphasized signal $s(n)$ to obtain the LP filter $A(z)$. Also, a new perceptual weighting filter 105 with fixed denominator is used. An example of transfer function for the perceptual weighting filter 104 is given by the following relation:

$$15 \quad W(z) = A(z/\gamma_1) / (1 - \gamma_2 z^{-1}) \quad \text{where} \quad 0 < \gamma_2 < \gamma_1 \leq 1$$

A higher order can be used at the denominator. This structure substantially decouples the formant weighting from the tilt.

20

Note that because $A(z)$ is computed based on the preemphasized speech signal $s(n)$, the tilt of the filter $1/A(z/\gamma_1)$ is less pronounced compared to the case when $A(z)$ is computed based on the original speech. Since deemphasis is performed at the decoder end using a filter having the transfer

25 function:

$$P^{-1}(z) = 1/(1 - \mu z^{-1}),$$

the quantization error spectrum is shaped by a filter having a transfer function $W^{-1}(z)P^{-1}(z)$. When γ_2 is set equal to μ , which is typically the case, the spectrum of the quantization error is shaped by a filter whose transfer

5 function is $1/A(z/\gamma_1)$, with $A(z)$ computed based on the preemphasized speech signal. Subjective listening showed that this structure for achieving the error shaping by a combination of preemphasis and modified weighting filtering is very efficient for encoding wideband signals, in addition to the advantages of ease of fixed-point algorithmic implementation.

10

Pitch Analysis:

In order to simplify the pitch analysis, an open-loop pitch lag T_{OL} is

15 first estimated in the open-loop pitch search module 106 using the weighted speech signal $s_w(n)$. Then the closed-loop pitch analysis, which is performed in closed-loop pitch search module 107 on a subframe basis, is restricted around the open-loop pitch lag T_{OL} which significantly reduces the search complexity of the LTP parameters T and b (pitch lag and pitch gain). Open-

20 loop pitch analysis is usually performed in module 106 once every 10 ms (two subframes) using techniques well known to those of ordinary skill in the art.

The target vector x for LTP (Long Term Prediction) analysis is first

25 computed. This is usually done by subtracting the zero-input response s_0 of weighted synthesis filter $W(z)/\hat{A}(z)$ from the weighted speech signal $s_w(n)$. This zero-input response s_0 is calculated by a zero-input response calculator

108. More specifically, the target vector x is calculated using the following relation:

$$x = s_w - s_0$$

5

where x is the N -dimensional target vector, s_w is the weighted speech vector in the subframe, and s_0 is the zero-input response of filter $W(z)/\hat{A}(z)$ which is the output of the combined filter $W(z)/\hat{A}(z)$ due to its initial states.

10 The zero-input response calculator 108 is responsive to the quantized interpolated LP filter $\hat{A}(z)$ from the LP analysis, quantization and interpolation calculator 104 and to the initial states of the weighted synthesis filter $W(z)/\hat{A}(z)$ stored in memory module 111 to calculate the zero-input response s_0 (that part of the response due to the initial states as determined by setting
15 the inputs equal to zero) of filter $W(z)/\hat{A}(z)$. This operation is well known to those of ordinary skill in the art and, accordingly, will not be further described.

Of course, alternative but mathematically equivalent approaches can be used to compute the target vector x .

20

A N -dimensional impulse response vector h of the weighted synthesis filter $W(z)/\hat{A}(z)$ is computed in the impulse response generator 109 using the LP filter coefficients $A(z)$ and $\hat{A}(z)$ from module 104. Again, this operation is well known to those of ordinary skill in the art and, accordingly,
25 will not be further described in the present specification.

The closed-loop pitch (or pitch codebook) parameters b , T and j are

computed in the closed-loop pitch search module 107, which uses the target vector x , the impulse response vector h and the open-loop pitch lag T_{ol} as inputs. Traditionally, the pitch prediction has been represented by a pitch filter having the following transfer function:

5

$$1 / (1 - bz^{-T})$$

10

where b is the pitch gain and T is the pitch delay or lag. In this case, the pitch contribution to the excitation signal $u(n)$ is given by $bu(n-T)$, where the total excitation is given by

15

$$u(n) = bu(n-T) + gc_k(n)$$

with g being the innovative codebook gain and $c_k(n)$ the innovative codevector at index k .

20

This representation has limitations if the pitch lag T is shorter than the subframe length N . In another representation, the pitch contribution can be seen as a pitch codebook containing the past excitation signal. Generally, each vector in the pitch codebook is a shift-by-one version of the previous vector (discarding one sample and adding a new sample). For pitch lags

25

$T > N$, the pitch codebook is equivalent to the filter structure $(1/(1-bz^{-T}))$, and a pitch codebook vector $v_T(n)$ at pitch lag T is given by

$$v_T(n) = u(n-T), \quad n=0, \dots, N-1.$$

For pitch lags T shorter than N , a vector $v_T(n)$ is built by repeating the available samples from the past excitation until the vector is completed (this is not equivalent to the filter structure).

In recent encoders, a higher pitch resolution is used which significantly improves the quality of voiced sound segments. This is achieved by oversampling the past excitation signal using polyphase interpolation filters. In this case, the vector $v_T(n)$ usually corresponds to an interpolated version of the past excitation, with pitch lag T being a non-integer delay (e.g. 50.25).

The pitch search consists of finding the best pitch lag T and gain b that minimize the mean squared weighted error E between the target vector x and the scaled filtered past excitation. Error E being expressed as:

$$E = \|x - by_T\|^2$$

where y_T is the filtered pitch codebook vector at pitch lag T :

$$y_T(n) = v_T(n) * h(n) = \sum_{i=0}^n v_T(i)h(n-i), \quad n=0, \dots, N-1.$$

It can be shown that the error E is minimized by maximizing the search criterion

5

$$C = \frac{x^t y_T}{\sqrt{y_T^t y_T}}$$

10 where t denotes vector transpose.

In the preferred embodiment of the present invention, a 1/3 subsample pitch resolution is used, and the pitch (pitch codebook) search is composed of three stages.

15

In the first stage, an open-loop pitch lag T_{OL} is estimated in open-loop pitch search module 106 in response to the weighted speech signal $s_w(n)$. As indicated in the foregoing description, this open-loop pitch analysis is usually performed once every 10 ms (two subframes) using techniques well
20 known to those of ordinary skill in the art.

In the second stage, the search criterion C is searched in the closed-loop pitch search module 107 for integer pitch lags around the estimated open-loop pitch lag T_{OL} (usually ± 5), which significantly simplifies the search
25 procedure. A simple procedure is used for updating the filtered codevector y_T without the need to compute the convolution for every pitch lag.

Once an optimum integer pitch lag is found in the second stage, a third stage of the search (module 107) tests the fractions around that optimum integer pitch lag.

When the pitch predictor is represented by a filter of the form $1/(1-bz^{-T})$, which is a valid assumption for pitch lags $T > N$, the spectrum of the pitch filter exhibits a harmonic structure over the entire frequency range, with a harmonic frequency related to $1/T$. In case of wideband signals, this structure is not very efficient since the harmonic structure in wideband signals does not cover the entire extended spectrum. The harmonic structure exists only up to a certain frequency, depending on the speech segment. Thus, in order to achieve efficient representation of the pitch contribution in voiced segments of wideband speech, the pitch prediction filter needs to have the flexibility of varying the amount of periodicity over the wideband spectrum.

15

A new method which achieves efficient modeling of the harmonic structure of the speech spectrum of wideband signals is disclosed in the present specification, whereby several forms of low pass filters are applied to the past excitation and the low pass filter with higher prediction gain is selected.

20

When subsample pitch resolution is used, the low pass filters can be incorporated into the interpolation filters used to obtain the higher pitch resolution. In this case, the third stage of the pitch search, in which the fractions around the chosen integer pitch lag are tested, is repeated for the several interpolation filters having different low-pass characteristics and the fraction and filter index which maximize the search criterion C are selected.

25

A simpler approach is to complete the search in the three stages described above to determine the optimum fractional pitch lag using only one interpolation filter with a certain frequency response, and select the optimum low-pass filter shape at the end by applying the different predetermined low-pass filters to the chosen pitch codebook vector v_T and select the low-pass filter which minimizes the pitch prediction error. This approach is discussed in detail below.

Figure 3 illustrates a schematic block diagram of a preferred embodiment of the proposed approach.

10

In memory module 303, the past excitation signal $u(n)$, $n < 0$, is stored. The pitch codebook search module 301 is responsive to the target vector x , to the open-loop pitch lag T_{OL} and to the past excitation signal $u(n)$, $n < 0$, from memory module 303 to conduct a pitch codebook (pitch codebook) search minimizing the above-defined search criterion C. From the result of the search conducted in module 301, module 302 generates the optimum pitch codebook vector v_T . Note that since a sub-sample pitch resolution is used (fractional pitch), the past excitation signal $u(n)$, $n < 0$, is interpolated and the pitch codebook vector v_T corresponds to the interpolated past excitation signal. In this preferred embodiment, the interpolation filter (in module 301, but not shown) has a low-pass filter characteristic removing the frequency contents above 7000 Hz.

20

In a preferred embodiment, K filter characteristics are used; these filter characteristics could be low-pass or band-pass filter characteristics. Once the optimum codevector v_T is determined and supplied by the pitch codevector generator 302, K filtered versions of v_T are computed

25

respectively using K different frequency shaping filters such as 305^(j), where $j=1, 2, \dots, K$. These filtered versions are denoted $v_f^{(j)}$, where $j=1, 2, \dots, K$. The different vectors $v_f^{(j)}$ are convolved in respective modules 304^(j), where $j=0, 1, 2, \dots, K$, with the impulse response h to obtain the vectors $y^{(j)}$, where $j=0, 1, 2, \dots, K$. To calculate the mean squared pitch prediction error for each vector $y^{(j)}$, the value $y^{(j)}$ is multiplied by the gain b by means of a corresponding amplifier 307^(j) and the value $by^{(j)}$ is subtracted from the target vector x by means of a corresponding subtractor 308^(j). Selector 309 selects the frequency shaping filter 305^(j) which minimizes the mean squared pitch prediction error

10

$$e^{(j)} = \|x - b^{(j)} y^{(j)}\|^2, \quad j=1, 2, \dots, K$$

15 To calculate the mean squared pitch prediction error $e^{(j)}$ for each value of $y^{(j)}$, the value $y^{(j)}$ is multiplied by the gain b by means of a corresponding amplifier 307^(j) and the value $b^{(j)} y^{(j)}$ is subtracted from the target vector x by means of subtractors 308^(j). Each gain $b^{(j)}$ is calculated in a corresponding gain calculator 306^(j) in association with the frequency shaping filter at index j , using the following relationship:

20

$$b^{(j)} = x^T y^{(j)} / \|y^{(j)}\|^2$$

25

In selector 309, the parameters b , T , and j are chosen based on v_T or $v_f^{(j)}$ which minimizes the mean squared pitch prediction error e .

Referring back to Figure 1, the pitch codebook index T is encoded and transmitted to multiplexer 112. The pitch gain b is quantized and transmitted to multiplexer 112. With this new approach, extra information is needed to encode the index j of the selected frequency shaping filter in multiplexer 112. For example, if three filters are used ($j=0, 1, 2, 3$), then two bits are needed to represent this information. The filter index information j can also be encoded jointly with the pitch gain b .

Innovative codebook search:

10

Once the pitch, or LTP (Long Term Prediction) parameters b , T , and j are determined, the next step is to search for the optimum innovative excitation by means of search module 110 of Figure 1. First, the target vector x is updated by subtracting the LTP contribution:

15

$$x' = x - by_T$$

where b is the pitch gain and y_T is the filtered pitch codebook vector (the past excitation at delay T filtered with the selected low pass filter and convolved with the impulse response h as described with reference to Figure 3).

25

The search procedure in CELP is performed by finding the optimum excitation codevector c_k and gain g which minimize the mean-squared error between the target vector and the scaled filtered codevector

$$E = \| x' - gHc_k \|^2$$

where H is a lower triangular convolution matrix derived from the impulse
5 response vector h .

In the preferred embodiment of the present invention, the innovative
codebook search is performed in module 110 by means of an algebraic
codebook as described in US patents Nos: 5,444,816 (Adoul et al.) issued
10 on August 22, 1995; 5,699,482 granted to Adoul et al., on December 17,
1997; 5,754,976 granted to Adoul et al., on May 19, 1998; and 5,701,392
(Adoul et al.) dated December 23, 1997.

Once the optimum excitation codevector c_k and its gain g are chosen
15 by module 110, the codebook index k and gain g are encoded and
transmitted to multiplexer 112.

Referring to Figure 1, the parameters b , T , j , $\hat{A}(z)$, k and g are
multiplexed through the multiplexer 112 before being transmitted through a
20 communication channel.

Memory update:

25 In memory module 111 (Figure 1), the states of the weighted
synthesis filter $W(z)/\hat{A}(z)$ are updated by filtering the excitation signal

$u = gc_k + bv_T$ through the weighted synthesis filter. After this filtering, the states of the filter are memorized and used in the next subframe as initial states for computing the zero-input response in calculator module 108.

5 As in the case of the target vector x , other alternative but mathematically equivalent approaches well known to those of ordinary skill in the art can be used to update the filter states.

DECODER SIDE

10

The speech decoding device 200 of Figure 2 illustrates the various steps carried out between the digital input 222 (input stream to the demultiplexer 217) and the output sampled speech 223 (output of the adder 221).

15

Demultiplexer 217 extracts the synthesis model parameters from the binary information received from a digital input channel. From each received binary frame, the extracted parameters are:

20

- the short-term prediction parameters (STP) $\hat{A}(z)$ (once per frame);

- the long-term prediction (LTP) parameters T , b , and j (for each subframe); and

25

- the innovation codebook index k and gain g (for each subframe).

The current speech signal is synthesized based on these parameters as will be explained hereinbelow.

5 The innovative codebook 218 is responsive to the index k to produce the innovation codevector $\mathbf{c}_k$, which is scaled by the decoded gain factor g through an amplifier 224. In the preferred embodiment, an innovative codebook 218 as described in the above mentioned US patent numbers 5,444,816; 5,699,482; 5,754,976; and 5,701,392 is used to represent the innovative codevector $\mathbf{c}_k$.

10 The generated scaled codevector $g\mathbf{c}_k$ at the output of the amplifier 224 is processed through an innovation filter 205.

Periodicity enhancement:

15 The generated scaled codevector at the output of the amplifier 224 is processed through a frequency-dependent pitch enhancer 205.

20 Enhancing the periodicity of the excitation signal $\mathbf{u}$ improves the quality in case of voiced segments. This was done in the past by filtering the innovation vector from the innovative codebook (fixed codebook) 218 through a filter in the form $1/(1-\varepsilon bz^{-T})$ where ε is a factor below 0.5 which controls the amount of introduced periodicity. This approach is less efficient in case of wideband signals since it introduces periodicity over the entire spectrum. A new alternative approach, which is part of the
25 present invention, is disclosed whereby periodicity enhancement is achieved by filtering the innovative codevector $\mathbf{c}_k$ from the innovative (fixed) codebook through an innovation filter 205 ($F(z)$) whose frequency

response emphasizes the higher frequencies more than lower frequencies. The coefficients of $F(z)$ are related to the amount of periodicity in the excitation signal u .

Many methods known to those skilled in the art are available for obtaining valid periodicity coefficients. For example, the value of gain b provides an indication of periodicity. That is, if gain b is close to 1, the periodicity of the excitation signal u is high, and if gain b is less than 0.5, then periodicity is low.

Another efficient way to derive the filter $F(z)$ coefficients used in a preferred embodiment, is to relate them to the amount of pitch contribution in the total excitation signal u . This results in a frequency response depending on the subframe periodicity, where higher frequencies are more strongly emphasized (stronger overall slope) for higher pitch gains. Innovation filter 205 has the effect of lowering the energy of the innovative codevector c_k at low frequencies when the excitation signal u is more periodic, which enhances the periodicity of the excitation signal u at lower frequencies more than higher frequencies. Suggested forms for innovation filter 205 are

$$(1) F(z) = 1 - \sigma z^{-1}, \quad \text{or} \quad (2) F(z) = -\alpha z + 1 - \alpha z^{-1}$$

where σ or α are periodicity factors derived from the level of periodicity of the excitation signal u .

The second three-term form of $F(z)$ is used in a preferred embodiment. The periodicity factor α is computed in the voicing factor generator 204. Several methods can be used to derive the periodicity factor α based on the periodicity of the excitation signal u . Two methods are presented below.

5

Method 1:

The ratio of pitch contribution to the total excitation signal u is first computed in voicing factor generator 204 by

10

$$R_p = \frac{b^2 \mathbf{v}_T^T \mathbf{v}_T}{\mathbf{u}^T \mathbf{u}} = \frac{b^2 \sum_{n=0}^{N-1} v_T^2(n)}{\sum_{n=0}^{N-1} u^2(n)}$$

15

where $\mathbf{v}_T$ is the pitch codebook vector, b is the pitch gain, and u is the excitation signal u given at the output of the adder 219 by

$$20 \quad \mathbf{u} = g\mathbf{c}_k + b\mathbf{v}_T$$

Note that the term $b\mathbf{v}_T$ has its source in the pitch codebook (pitch codebook) 201 in response to the pitch lag T and the past value of u stored in memory 203. The pitch codevector $\mathbf{v}_T$ from the pitch codebook 201 is then processed through a low-pass filter 202 whose cut-off frequency is adjusted by means of the index j from the demultiplexer 217. The resulting codevector $\mathbf{v}_T$ is then multiplied by the gain b from the

25

demultiplexer 217 through an amplifier 226 to obtain the signal bv_T .

The factor α is calculated in voicing factor generator 204 by

$$\alpha = qR_p \quad \text{bounded by} \quad \alpha < q$$

5

where q is a factor which controls the amount of enhancement (q is set to 0.25 in this preferred embodiment).

Method 2:

10

Another method used in a preferred embodiment of the invention for calculating periodicity factor α is discussed below.

First, a voicing factor r_v is computed in voicing factor generator 204 by

15

$$r_v = (E_v - E_c) / (E_v + E_c)$$

20 where E_v is the energy of the scaled pitch codevector bv_T and E_c is the energy of the scaled innovative codevector gc_k . That is

$$E_v = b^2 v_T^t v_T = b^2 \sum_{n=0}^{N-1} v_T^2(n)$$

25 and

$$E_c = g^2 c_k^t c_k = g^2 \sum_{n=0}^{N-1} c_k^2(n)$$

Note that the value of r_v lies between -1 and 1 (1 corresponds to purely voiced signals and -1 corresponds to purely unvoiced signals).

In this preferred embodiment, the factor α is then computed in voicing factor generator 204 by

5

$$\alpha = 0.125 (1 + r_v)$$

10 which corresponds to a value of 0 for purely unvoiced signals and 0.25 for purely voiced signals.

In the first, two-term form of $F(z)$, the periodicity factor σ can be approximated by using $\sigma = 2\alpha$ in methods 1 and 2 above. In such a case, the periodicity factor σ is calculated as follows in method 1 above:

15 $\sigma = 2qR_p$ bounded by $\sigma < 2q$.

In method 2, the periodicity factor σ is calculated as follows:

20 $\sigma = 0.25 (1 + r_v)$.

The enhanced signal c_r is therefore computed by filtering the scaled innovative codevector gc_k through the innovation filter 205 ($F(z)$).

The enhanced excitation signal u' is computed by the adder 220

25 as:

$$u' = c_r + bv_T$$

Note that this process is not performed at the encoder 100. Thus, it is essential to update the content of the pitch codebook 201 using the excitation signal u without enhancement to keep synchronism between the encoder 100 and decoder 200. Therefore, the excitation signal u is used to update the memory 203 of the pitch codebook 201 and the enhanced excitation signal u' is used at the input of the LP synthesis filter 206.

Synthesis and deemphasis

The synthesized signal s' is computed by filtering the enhanced excitation signal u' through the LP synthesis filter 206 which has the form $1/\hat{A}(z)$, where $\hat{A}(z)$ is the interpolated LP filter in the current subframe. As can be seen in Figure 2, the quantized LP coefficients $\hat{A}(z)$ on line 225 from demultiplexer 217 are supplied to the LP synthesis filter 206 to adjust the parameters of the LP synthesis filter 206 accordingly. The deemphasis filter 207 is the inverse of the preemphasis filter 103 of Figure 1. The transfer function of the deemphasis filter 207 is given by

$$D(z) = 1 / (1 - \mu z^{-1})$$

where μ is a preemphasis factor with a value located between 0 and 1 (a typical value is $\mu = 0.7$). A higher-order filter could also be used.

The vector $\mathbf{s}'$ is filtered through the deemphasis filter $D(z)$ (module 207) to obtain the vector $\mathbf{s}_b$ which is passed through the high-pass filter 208
5 to remove the unwanted frequencies below 50 Hz and further obtain $\mathbf{s}_h$.

Oversampling and high-frequency regeneration

10 The over-sampling module 209 conducts the inverse process of the down-sampling module 101 of Figure 1. In this preferred embodiment, oversampling converts from the 12.8 kHz sampling rate to the original 16 kHz sampling rate, using techniques well known to those of ordinary skill in the art. The oversampled synthesis signal is denoted $\hat{\mathbf{S}}$. Signal $\hat{\mathbf{S}}$ is also
15 referred to as the synthesized wideband intermediate signal.

The oversampled synthesis $\hat{\mathbf{S}}$ signal does not contain the higher frequency components which were lost by the downsampling process (module 101 of Figure 1) at the encoder 100. This gives a low-pass
20 perception to the synthesized speech signal. To restore the full band of the original signal, a high frequency generation procedure is disclosed. This procedure is performed in modules 210 to 216, and adder 221, and requires input from voicing factor generator 204 (Figure 2).

25 In this new approach, the high frequency contents are generated by filling the upper part of the spectrum with a white noise properly scaled in the excitation domain, then converted to the speech domain, preferably by

shaping it with the same LP synthesis filter used for synthesizing the down-sampled signal $\hat{s}$.

The high frequency generation procedure in accordance with the present invention is described hereinbelow.

5

The random noise generator 213 generates a white noise sequence w' with a flat spectrum over the entire frequency bandwidth, using techniques well known to those of ordinary skill in the art. The generated sequence is of length N' which is the subframe length in the original domain.

10 Note that N is the subframe length in the down-sampled domain. In this preferred embodiment, $N=64$ and $N'=80$ which correspond to 5 ms.

The white noise sequence is properly scaled in the gain adjusting module 214. Gain adjustment comprises the following steps. First, the energy of the generated noise sequence w' is set equal to the energy of the enhanced excitation signal u' computed by an energy computing module 210, and the resulting scaled noise sequence is given by

15

20

$$w(n) = w'(n) \sqrt{\frac{\sum_{n=0}^{N-1} u'^2(n)}{\sum_{n=0}^{N'-1} w'^2(n)}}, \quad n=0, \dots, N'-1.$$

25

The second step in the gain scaling is to take into account the high frequency contents of the synthesized signal at the output of the voicing factor generator 204 so as to reduce the energy of the generated noise in

case of voiced segments (where less energy is present at high frequencies compared to unvoiced segments). In this preferred embodiment, measuring the high frequency contents is implemented by measuring the tilt of the synthesis signal through a spectral tilt calculator 212 and reducing the energy accordingly. Other measurements such as zero crossing measurements can equally be used. When the tilt is very strong, which corresponds to voiced segments, the noise energy is further reduced. The tilt factor is computed in module 212 as the first correlation coefficient of the synthesis signal s_h and it is given by:

$$tilt = \frac{\sum_{n=1}^{N-1} s_h(n) s_h(n-1)}{\sum_{n=0}^{N-1} s_h^2(n)}, \quad \text{conditioned by } tilt \geq 0 \text{ and } tilt \geq r_v$$

where voicing factor r_v is given by

$$r_v = (E_v - E_c) / (E_v + E_c)$$

where E_v is the energy of the scaled pitch codevector bv_r and E_c is the energy of the scaled innovative codevector gc_n as described earlier. Voicing factor r_v is most often less than $tilt$ but this condition was introduced as a

precaution against high frequency tones where the tilt value is negative and the value of r_v is high. Therefore, this condition reduces the noise energy for such tonal signals.

5 The tilt value is 0 in case of flat spectrum and 1 in case of strongly voiced signals, and it is negative in case of unvoiced signals where more energy is present at high frequencies.

Different methods can be used to derive the scaling factor g_t from the amount of high frequency contents. In this invention, two methods are given
10 based on the tilt of signal described above.

Method 1:

15 The scaling factor g_t is derived from the tilt by

$$g_t = 1 - \text{tilt} \quad \text{bounded by} \quad 0.2 \leq g_t \leq 1.0$$

20 For strongly voiced signal where the tilt approaches 1, g_t is 0.2 and for strongly unvoiced signals g_t becomes 1.0.

Method 2:

25 The tilt factor g_t is first restricted to be larger or equal to zero, then the scaling factor is derived from the tilt by

$$g_t = 10^{-0.6 \text{tilt}}$$

The scaled noise sequence w_g produced in gain adjusting module 214 is therefore given by:

5

$$w_g = g_t w.$$

When the tilt is close to zero, the scaling factor g_t is close to 1, which does not result in energy reduction. When the tilt value is 1, the scaling factor g_t results in a reduction of 12 dB in the energy of the generated noise.

10

Once the noise is properly scaled (w_g), it is brought into the speech domain using the spectral shaper 215. In the preferred embodiment, this is achieved by filtering the noise w_g through a bandwidth expanded version of the same LP synthesis filter used in the down-sampled domain ($1/\hat{A}(z/0.8)$). The corresponding bandwidth expanded LP filter coefficients are calculated in spectral shaper 215.

15

The filtered scaled noise sequence w_f is then band-pass filtered to the required frequency range to be restored using the band-pass filter 216. In the preferred embodiment, the band-pass filter 216 restricts the noise sequence to the frequency range 5.6-7.2 kHz. The resulting band-pass filtered noise sequence z is added in adder 221 to the oversampled synthesized speech signal $\hat{s}$ to obtain the final reconstructed sound signal s_{out} on the output 223.

20

25

Although the present invention has been described hereinabove by way of a preferred embodiment thereof, this embodiment can be modified at will, within the scope of the appended claims, without departing from the spirit and nature of the subject invention. Even though the preferred embodiment discusses the use of wideband speech signals, it will be
5 obvious to those skilled in the art that the subject invention is also directed to other embodiments using wideband signals in general and that it is not necessarily limited to speech applications.

WHAT IS CLAIMED IS:

1. A pitch analysis device for producing an optimal set of pitch codebook parameters, comprising:
 - a) at least two signal paths associated to respective sets of pitch codebook parameters, wherein:
 - i) each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device; and
 - ii) at least one of said two paths comprises a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of said one path; and
 - b) a selector for comparing the pitch prediction errors calculated in said at least two signal paths, for choosing the signal path having the lowest calculated pitch prediction error, and for selecting the set of pitch codebook parameters associated to the chosen signal path.
2. A pitch analysis device as defined in claim 1, wherein one of said at least two paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.
3. A pitch analysis device as defined in claim 1, wherein said signal paths comprises a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same path.

4. A pitch analysis device as defined in claim 3, wherein the filters of said plurality of paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

5. A pitch analysis device as defined in claim 1, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with a weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

6. A pitch analysis device as defined in claim 5, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = \frac{x^T y^{(j)}}{\|y^{(j)}\|^2} \quad \text{where } j = 0, 1, 2, \dots, K, \text{ and } K \text{ corresponds to a number of signal paths,}$$

and where x is said pitch search target vector, and $y^{(j)}$ is said convolved pitch codevector.

7. A pitch analysis device as defined in claim 1, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the
5 signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

8. A pitch analysis device as defined in claim 5, wherein:
a) each of said filters of the plurality of signal paths is identified by a filter
10 index;
b) said pitch codevector is identified by a pitch codebook index; and
c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

15 9. A pitch analysis device as defined in claim 1, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

20 10. A pitch analysis method for producing an optimal set of pitch codebook parameters, comprising:
a) in at least two signal paths associated to respective sets of pitch codebook parameters, calculating, for each signal path, a pitch prediction error of a pitch codevector from a pitch codebook search device;
25 b) in at least one of said two signal paths, filtering the pitch codevector before supplying said pitch codevector for calculation of said pitch prediction error of said one path; and

c) comparing the pitch prediction errors calculated in said at least two signal paths, choosing the signal path having the lowest calculated pitch prediction error, and selecting the set of pitch codebook parameters associated to the chosen signal path.

5 11. A pitch analysis method as defined in claim 10, wherein, in one of said at least two paths, no filtering of the pitch codevector is performed before supplying said pitch codevector to the pitch prediction error calculating device.

10 12. A pitch analysis method as defined in claim 10, wherein said signal paths comprises a plurality of signal paths and wherein filtering the pitch codevector is performed in each of said plurality of signal paths before supplying said pitch codevector to the pitch prediction error calculating device of the same path.

15 13. A pitch analysis method as defined in claim 12, further comprising selecting the filters of said plurality of paths from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

20 14. A pitch analysis method as defined in claim 10, wherein calculating a pitch prediction error in each signal path comprises:

25 a) convolving the pitch codevector with a weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

 b) calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;

c) multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

- 5 15. A pitch analysis method as defined in claim 14, wherein said pitch gain calculation comprises calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^T y^{(j)} / \|y^{(j)}\|^2 \quad \text{where } j = 0, 1, 2, \dots, K, \text{ and } K \text{ corresponds to a number of signal paths,}$$

10

and where x is said pitch search target vector, and $y^{(j)}$ is said convolved pitch codevector.

- 15 16. A pitch analysis method as defined in claim 10, wherein calculating said pitch prediction error, in each signal path, comprises calculating an energy of the corresponding pitch prediction error, and wherein comparing the pitch prediction error comprises comparing the energies of said pitch prediction errors of the different signal paths and choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest
20 calculated energy of the pitch prediction error.

17. A pitch analysis method as defined in claim 14, wherein:

- a) identifying each of said filters of the plurality of signal paths by a filter index;
25 b) identifying said pitch codevector by a pitch codebook index; and
c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

18. A pitch analysis method as defined in claim 10, wherein said filtering the pitch codevector is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

5 19. An encoder having a pitch analysis device as in claim 1 for encoding a wideband input signal, said encoder comprising:

a) a linear prediction synthesis filter calculator responsive to the wideband signal for producing linear prediction synthesis filter coefficients;

10 b) a perceptual weighting filter, responsive to the wideband signal and the linear prediction synthesis filter coefficients, for producing a perceptually weighted signal;

c) an impulse response generator responsive to said linear prediction synthesis filter coefficients for producing a weighted synthesis filter impulse response signal;

15 d) a pitch search unit for producing pitch codebook parameters, said pitch search unit comprising:

20 i) said pitch codebook search device responsive to the perceptually weighted signal and the linear prediction synthesis filter coefficients for producing the pitch codevector and an innovative search target vector; and

ii) said pitch analysis device responsive to the pitch codevector for selecting, from said sets of pitch codebook parameters, the set of pitch codebook parameters associated to the path having the lowest calculated pitch prediction error;

25 d) an innovative codebook search device, responsive to the weighted synthesis filter impulse response signal, and the innovative search target vector, for producing innovative codebook parameters; and

e) a signal forming device for producing an encoded wideband signal comprising the set of pitch codebook parameters associated to the path having the lowest pitch prediction error, said innovative codebook parameters, and said linear prediction synthesis filter coefficients.

5 20. An encoder as defined in claim 19, wherein one of said at least two paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

10 21. An encoder as defined in claim 19, wherein said signal paths comprises a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same path.

15 22. An encoder as defined in claim 21, wherein the filters of said plurality of paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

20 23. An encoder as defined in claim 19, wherein each pitch prediction error calculating device comprises:

20 a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

 b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and the pitch search target vector;

25 c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

24. An encoder as defined in claim 23, wherein said pitch gain calculator
5 comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^T y^{(j)} / \|y^{(j)}\|^2 \quad \text{where } j = 0, 1, 2, \dots, K, \text{ and } K \text{ corresponds to a number of signal paths,}$$

10 and where x is said pitch search target vector, and $y^{(j)}$ is said convolved pitch codevector.

25. An encoder as defined in claim 19, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an
15 energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

20

26. An encoder as defined in claim 23, wherein:

a) each of said filters of the plurality of signal paths is identified by a filter index;

b) said pitch codevector is identified by a pitch codebook index; and

25 c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

27. A encoder as defined in claim 19, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

28. A cellular communication system for servicing a large geographical area
5 divided into a plurality of cells, comprising:

- a) mobile transmitter/receiver units;
- b) cellular base stations respectively situated in said cells;
- c) a control terminal for controlling communication between the cellular base stations;

10 d) a bidirectional wireless communication sub-system between each mobile unit situated in one cell and the cellular base station of said one cell, said bidirectional wireless communication sub-system comprising, in both the mobile unit and the cellular base station:

- 15 i) a transmitter including an encoder for encoding a wideband signal as recited in claim 19 and a transmission circuit for transmitting the encoded wideband signal; and
- ii) a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal.

20

29. A cellular communication system as defined in claim 28, wherein one of said at least two paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

25

30. A cellular communication system as defined in claim 28, wherein said signal paths comprises a plurality of signal paths each provided with a filter

for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same path.

31. A cellular communication system as defined in claim 30, wherein the filters of said plurality of paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

32. A cellular communication system as defined in claim 28, wherein each pitch prediction error calculating device comprises:

- a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;
- b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and the pitch search target vector;
- c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and
- d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

33. A cellular communication system as defined in claim 32, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^T y^{(j)} / \|y^{(j)}\|^2 \quad \text{where } j = 0, 1, 2, \dots, K, \text{ and } K \text{ corresponds to a number of signal paths,}$$

and where x is said pitch search target vector, and $y^{(j)}$ is said convolved pitch codevector.

34. A cellular communication system as defined in claim 28, wherein said pitch prediction error calculating device of each signal path comprises
5 means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

10

35. A cellular communication system as defined in claim 32, wherein:

a) each of said filters of the plurality of signal paths is identified by a filter index;

b) said pitch codevector is identified by a pitch codebook index; and

15

c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

36. A cellular communication system as defined in claim 28, wherein said filter is integrated in an interpolation filter of said pitch codebook search
20 device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

37. A cellular mobile transmitter/receiver unit comprising:

a) a transmitter including an encoder for encoding a wideband
25 signal as recited in claim 19 and a transmission circuit for transmitting the encoded wideband signal; and

b) a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal.

5 38. A cellular mobile transmitter/receiver unit as defined in claim 37, wherein one of said at least two paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

10 39. A cellular mobile transmitter/receiver unit as defined in claim 37, wherein said signal paths comprises a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same path.

15 40. A cellular mobile transmitter/receiver unit as defined in claim 39, wherein the filters of said plurality of paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

20 41. A cellular mobile transmitter/receiver unit as defined in claim 37, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

25 b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and the pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

42. A cellular mobile transmitter/receiver unit as defined in claim 41, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = \mathbf{x}^T \mathbf{y}^{(j)} / \|\mathbf{y}^{(j)}\|^2 \quad \text{where } j = 0, 1, 2, \dots, K, \text{ and } K \text{ corresponds to a number of signal paths,}$$

and where $\mathbf{x}$ is said pitch search target vector, and $\mathbf{y}^{(j)}$ is said convolved pitch codevector.

43. A cellular mobile transmitter/receiver unit as defined in claim 37, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

44. A cellular mobile transmitter/receiver unit as defined in claim 41, wherein:

a) each of said filters of the plurality of signal paths is identified by a filter index;

b) said pitch codevector is identified by a pitch codebook index;
and

c) said pitch codebook parameters comprise the filter index, the
5 pitch codebook index and the pitch gain.

45. A cellular mobile transmitter/receiver unit as defined in claim 37, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version
10 of said pitch codevector.

46. A cellular network element comprising:

a) a transmitter including an encoder for encoding a wideband signal as recited in claim 19 and a transmission circuit for transmitting the
15 encoded wideband signal; and

b) a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal.

20 47. A cellular network element as defined in claim 46, wherein one of said at least two paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

25 48. A cellular network element as defined in claim 46, wherein said signal paths comprises a plurality of signal paths each provided with a filter for

filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same path.

49. A cellular network element as defined in claim 48, wherein the filters of said plurality of paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

50. A cellular network element as defined in claim 46, wherein each pitch prediction error calculating device comprises:

10 a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

 b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and the pitch search target vector;

15 c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

 d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

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51. A cellular network element as defined in claim 50, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

25
$$b^{(j)} = x^{(j)} y^{(j)} / \|y^{(j)}\|^2$$
 where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths,

and where x is said pitch search target vector, and $y^{(j)}$ is said convolved pitch codevector.

52. A cellular network element as defined in claim 46, wherein said pitch
5 prediction error calculating device of each signal path comprises means for
calculating an energy of the corresponding pitch prediction error, and
wherein said selector comprises means for comparing the energies of said
pitch prediction errors of the different signal paths and for choosing as the
signal path having the lowest calculated pitch prediction error the signal path
10 having the lowest calculated energy of the pitch prediction error.

53. A cellular network element as defined in claim 50, wherein:

- a) each of said filters of the plurality of signal paths is identified by
a filter index;
- 15 b) said pitch codevector is identified by a pitch codebook index;
and
- c) said pitch codebook parameters comprise the filter index, the
pitch codebook index and the pitch gain.

20 54. A cellular network element as defined in claim 46, wherein said filter is
integrated in an interpolation filter of said pitch codebook search device, said
interpolation filter being used to produce a sub-sample version of said pitch
codevector.

25 55. In a cellular communication system for servicing a large geographical
area divided into a plurality of cells, comprising: mobile transmitter/receiver

units; cellular base stations, respectively situated in said cells; and control terminal for controlling communication between the cellular base stations:

a bidirectional wireless communication sub-system between each mobile unit situated in one cell and the cellular base station of said one cell, said bidirectional wireless communication sub-system comprising, in both the

5 mobile unit and the cellular base station:

a) a transmitter including an encoder for encoding a wideband signal as recited in claim 19 and a transmission circuit for transmitting the encoded wideband signal; and

b) a receiver including a receiving circuit for receiving a transmitted

10 encoded wideband signal and a decoder for decoding the received encoded wideband signal.

56. A bidirectional wireless communication sub-system as defined in claim 55, wherein one of said at least two paths comprises no filter for filtering the

15 pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

57. A bidirectional wireless communication sub-system as defined in claim 55, wherein said signal paths comprises a plurality of signal paths each

20 provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same path.

58. A bidirectional wireless communication sub-system as defined in claim

25 57, wherein the filters of said plurality of paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

59. A bidirectional wireless communication sub-system as defined in claim 55, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

5 b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and the pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

10 d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

60. A bidirectional wireless communication sub-system as defined in claim 15 59, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^T y^{(j)} / \|y^{(j)}\|^2 \quad \text{where } j = 0, 1, 2, \dots, K, \text{ and } K \text{ corresponds to a number of signal paths,}$$

20

and where x is said pitch search target vector, and $y^{(j)}$ is said convolved pitch codevector.

61. A bidirectional wireless communication sub-system as defined in claim 25 55, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing

the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

5 62. A bidirectional wireless communication sub-system as defined in claim 59, wherein:

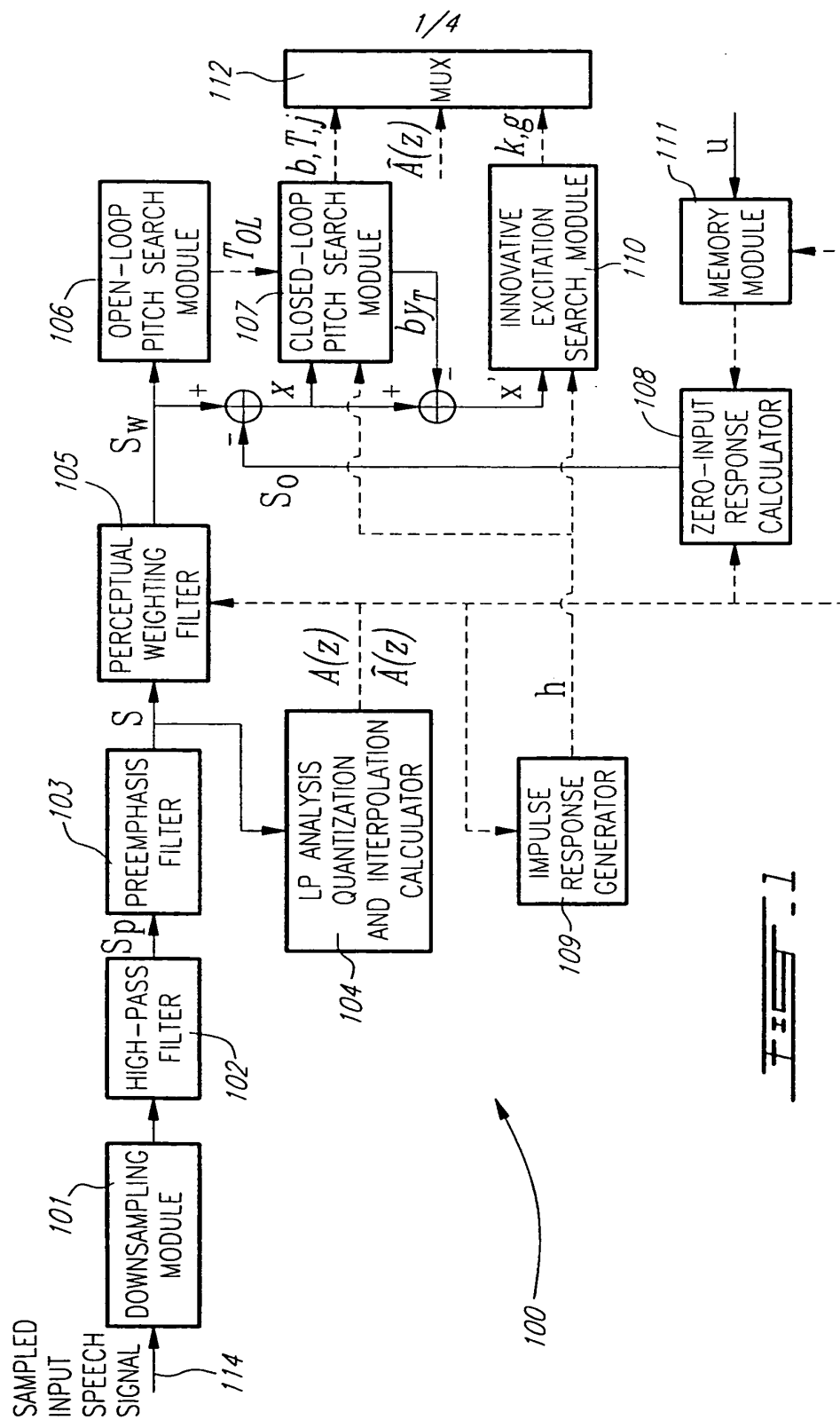
a) each of said filters of the plurality of signal paths is identified by a filter index;

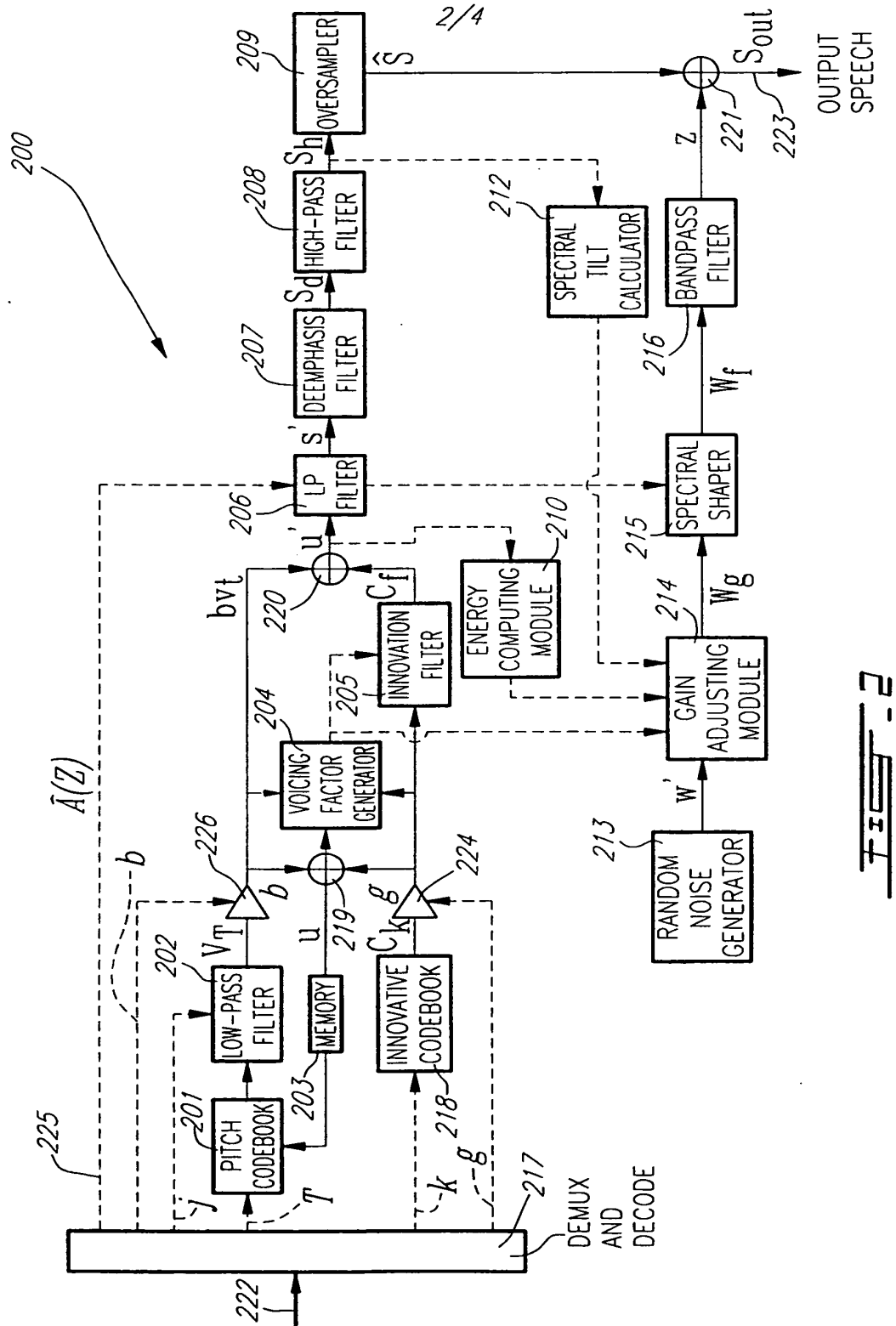
b) said pitch codevector is identified by a pitch codebook index;

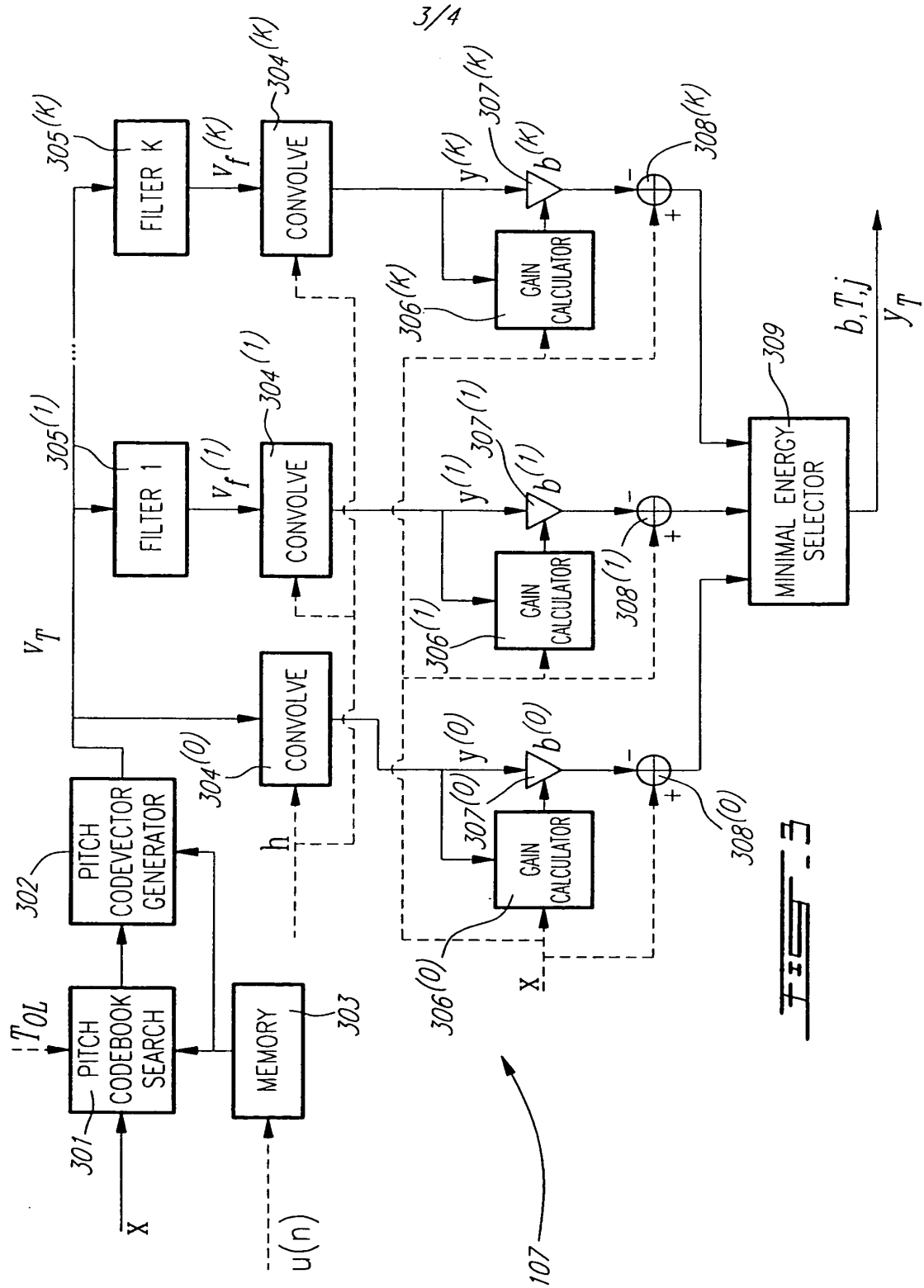
10 and

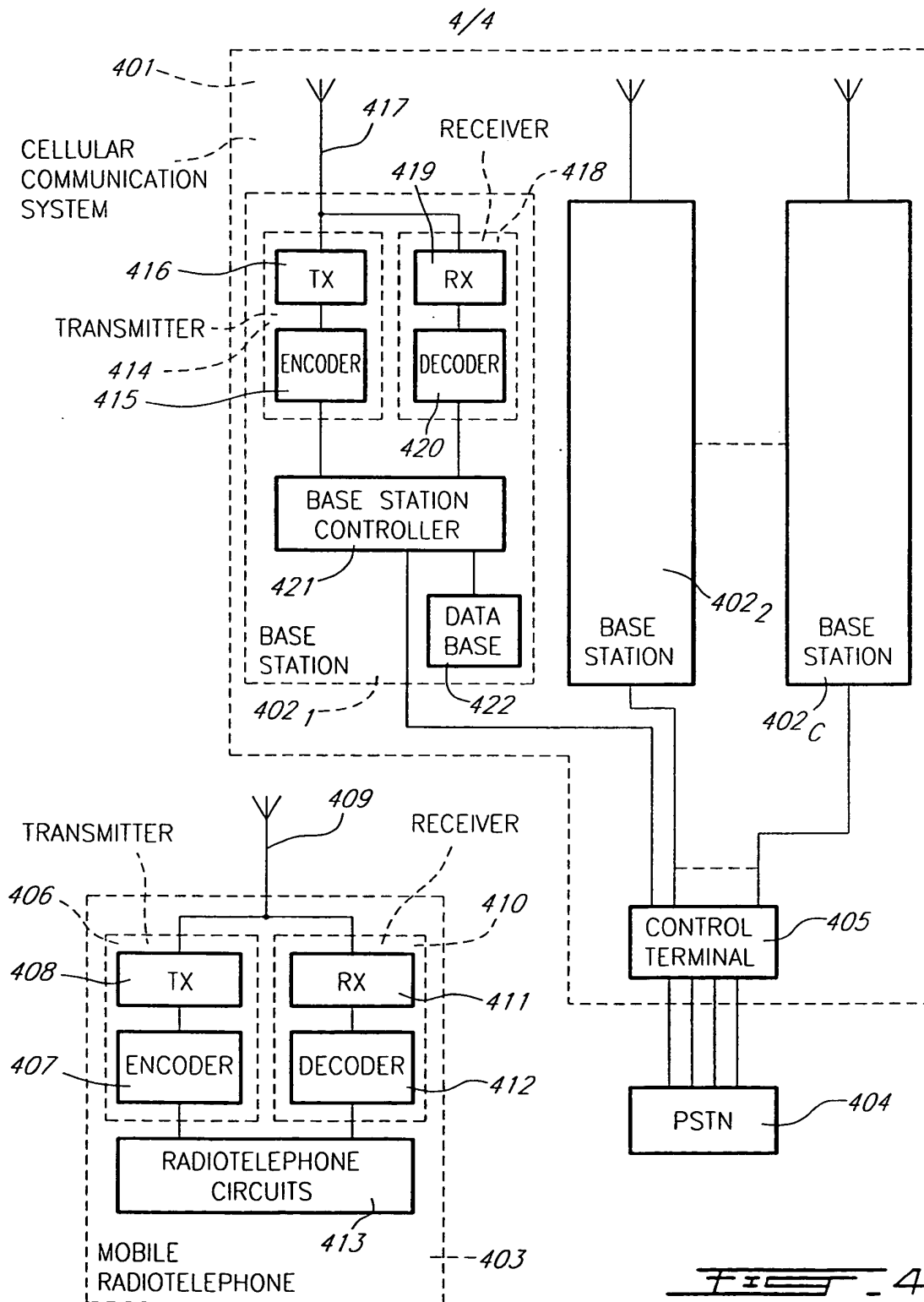
c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

15 63. A bidirectional wireless communication sub-system as defined in claim 55, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.









INTERNATIONAL SEARCH REPORT

International Application No

PCT/CA 99/01008

A. CLASSIFICATION OF SUBJECT MATTER

IPC 7 G10L11/04

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC 7 G10L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

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A	EP 0 331 858 A (IBM) 13 September 1989 (1989-09-13) abstract; claims 5-7 page 3, line 16 - line 25 page 9, line 35 - line 43 ----	1, 10
A	EP 0 722 165 A (DIGITAL VOICE SYSTEMS INC) 17 July 1996 (1996-07-17) abstract ----	1, 10
A	EP 0 421 444 A (FUJITSU LTD) 10 April 1991 (1991-04-10) abstract; claim 1 -----	1, 10

☐ Further documents are listed in the continuation of box C.

☒ Patent family members are listed in annex.

* Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

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"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

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Date of the actual completion of the international search

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INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No

PCT/CA 99/01008

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Specification and Drawings, as originally filed, with Application for Patent Serial No:
2,252,170, on October 27, 1998, by UNIVERSITE DE SHERBROOKE, assignee of
Bruno Bessette and Roch Lefebvre, for "A Method and Device for High Quality Coding
of Wideband Speech and Audio Signals"

PRIORITY DOCUMENT

SUBMITTED OR TRANSMITTED IN
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S. Gregoire
Agent certificateur/Certifying Officer

November 15, 1999

Date

Canada

(CIPO 68)

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A METHOD AND DEVICE FOR HIGH QUALITY CODING
OF WIDEBAND SPEECH AND AUDIO SIGNALS

5

BACKGROUND OF THE INVENTION

1. Field of the invention:

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The present invention relates to an efficient technique for digitally encoding a wideband sound signal, in particular but not exclusively a speech signal, in view of transmitting, or storing, and synthesizing this wideband sound signal.

15

2. Brief description of the prior art:

20 The demand for efficient digital wideband speech/audio encoding techniques with a good subjective quality/bit rate trade-off is increasing for numerous applications such as audio/video teleconferencing, multimedia, and wireless applications, as well as Internet and packet network applications. Until recently, telephone bandwidths filtered in the range 200-3400 Hz were mainly used in speech
25 coding applications. However, there is an increasing demand for wideband speech applications in order to increase the intelligibility and naturalness of the speech signals. A bandwidth in the range 50-7000 Hz was found sufficient for delivering a face-to-face speech quality. For

audio signals, this range gives an acceptable audio quality, but still lower than the CD quality which operates on the range 20-20000 Hz.

A speech encoder converts a speech signal into a digital bitstream which is transmitted over a communication channel (or stored in a storage medium). The speech signal is digitized (sampled and quantized with usually 16-bits per sample) and the speech encoder has the role of representing these digital samples with a smaller number of bits while maintaining a good subjective speech quality. The speech decoder or synthesizer operates on the transmitted or stored bit stream and converts it back to a sound signal.

One of the best prior art techniques capable of achieving a good quality/bit rate trade-off is the so-called Code Excited Linear Prediction (CELP) technique. According to this technique, the sampled speech signal is processed in successive blocks of L samples usually called *frames* where L is some predetermined number (corresponding to 10-30 ms of speech). In CELP, a linear prediction (LP) filter is computed and transmitted every frame. The L -sample frame is then divided into smaller blocks called *subframes* of size N samples, where $L=kN$ and k is the number of subframes in a frame (N usually corresponds to 4-10 ms of speech). An excitation signal is determined in each subframe, which usually consists of two components: one from the past excitation (also called pitch contribution or adaptive codebook) and the other from an innovation codebook (also called fixed codebook). This excitation signal is transmitted and used at the decoder as the input of the LP synthesis filter in order to obtain the synthesized speech.

An innovation codebook in the CELP context, is an indexed set of N -sample-long sequences which will be referred to as N -dimensional codevectors. Each codebook sequence is indexed by an integer k ranging from 1 to M where M represents the size of the codebook often expressed as a number of bits b , where $M=2^b$.

5

To synthesize speech according to the CELP technique, each block of N samples is synthesized by filtering an appropriate codevector from a codebook through time varying filters modeling the spectral characteristics of the speech signal. At the encoder end, the synthetic output is computed for all, or a subset, of the codevectors from the codebook (codebook search). The retained codevector is the one producing the synthetic output closest to the original speech signal according to a perceptually weighted distortion measure. This perceptual weighting is performed using a so-called perceptual weighting filter, which is usually derived from the LP filter.

10
15

The CELP model has been very successful in encoding telephone band sound signals, and several CELP-based standards exist in a wide range of applications, especially in digital cellular applications. In the telephone band, the sound signal is band-limited to 200-3400 Hz and sampled at 8000 samples/sec. In wideband speech/audio applications, the sound signal is band-limited to 50-7000 Hz and sampled at 16000 samples/sec.

20

Some difficulties arise when applying the telephone-band optimized CELP model to wideband signals, and additional features need to be added to the model in order to obtain high quality wideband signals. Wideband signals exhibit a much wider dynamic range compared to

25

telephone-band signals, which results in precision problems when a fixed-point implementation of the algorithm is required (which is essential in wireless applications). Further, the CELP model will often spend most of its encoding bits on the low-frequency region, which usually has higher energy contents, resulting in a low-pass output signal. To overcome this problem,

5 the perceptual weighting filter has to be modified in order to suit wideband signals, and pre-emphasis techniques which boost the high frequency regions become important to reduce the dynamic range, yielding a simpler fixed-point implementation, and to ensure a better encoding of the higher frequency contents of the signal. Further, the pitch contents in the spectrum

10 of voiced segments in wideband signals do not extend over the whole spectrum range, and the amount of voicing shows more variation compared to narrow-band signals. Thus, it is important to improve the closed-loop pitch analysis to better accommodate the variations in the voicing level.

15 At the decoder side, the CELP model uses post-filtering and post-processing techniques in order to improve the perceived synthesized signal. These techniques have to be changed to accommodate wideband signals. Further, in order to lower the bit rate below 16 kbit/s, an efficient method is to down-sample the wideband signals, which enables the encoder to operate

20 on a bandwidth lower than 7000 Hz, thus achieving a reduction in the bit rate. At the decoder side, the decoder signal is upsampled and an efficient high frequency generation technique is needed to recover the full band signal, while maintaining a quality close to the original signal.

25

OBJECTS OF THE INVENTION

An object of the present invention is therefore to provide a method and device for efficiently encoding wideband (7000 Hz) sound signals using
5 CELP-type encoding techniques, using additional features at both encoder and decoder in order to obtain high a quality reconstructed sound signal, which is also suitable for fixed point algorithmic implementation.

10

SUMMARY OF THE INVENTION

More specifically, in accordance with the present invention, there is provided a method for encoding wideband sound signals using LP-based,
15 preferably CELP-type encoding techniques, whereby the following new features are adopted in order to obtain high subjective quality of the decoded wideband sound signal:

1. The overall perceptual weighting of the quantization error is
20 obtained by a combination of a preemphasis filter and a modified weighting filter.

In CELP-type coders, the optimum pitch and innovation parameters are searched by minimizing the mean squared error between the
25 input speech and synthesized speech in a perceptually weighted domain. This is equivalent to minimizing the error between the weighted input speech and weighted synthesis speech, where the weighting is performed using a filter having a transfer function $W(z)$ of the form:

$$W(z) = A(z/\gamma_1)/A(z/\gamma_2) \text{ where } 0 < \gamma_2 < \gamma_1 \leq 1.$$

In analysis-by-synthesis (AbS) coders, analysis shows that the quantization error is weighted by the inverse of the weighting filter, $W^{-1}(z)$, which exhibits some of the formant structure in the input signal. Thus, the masking property of the human ear is exploited by shaping the error, so that it has more energy in the formant regions, where it will be masked by the strong signal energy present in those regions. The amount of weighting is controlled by the factors γ_1 and γ_2 .

This filter works well with telephone band signals. However, it was found that this filter is not suitable for efficient perceptual weighting when it was applied to wideband signals. It was found that this filter has inherent limitations in modeling the formant structure and the required spectral tilt concurrently. The spectral tilt is more pronounced in wideband signals due to the wide dynamic range between low and high frequencies. It was suggested to add a tilt filter into filter $W(z)$ in order to control the tilt and formant weighting separately.

A novel solution to this problem, forming part of the present invention, is to introduce a preemphasis filter at the input, compute the LP filter $A(z)$ based on the preemphasized speech, and use a modified filter $W(z)$ by fixing its denominator.

The preemphasis filter reduces the dynamic range of the input signal, which renders it more suitable for fixed-point implementation, and improves the encoding of the high frequency contents of the spectrum. The

preemphasis is obtained by a fixed FIR filter having a transfer function $P(z)$ in the form:

$$P(z) = 1 - \mu z^{-1}$$

5 where μ is a preemphasis factor with a value between 0 and 1. A higher order filter can also be used. Linear prediction (LP) analysis is performed on the preemphasized input signal to obtain the LP filter $A(z)$. A new weighting filter is used, which has a transfer function of the form:

$$10 \quad W(z) = A(z/\gamma_1)/(1 - \gamma_2 z^{-1}) \quad \text{where } 0 < \gamma_2 < \gamma_1 \leq 1$$

Note that because $A(z)$ is computed based on preemphasized speech, the tilt of the filter $1/A(z/\gamma_1)$ is less pronounced compared to the case when $A(z)$ is computed based on the original speech. Since deemphasis

15 using the filter $P^{-1}(z) = 1/(1 - \mu z^{-1})$ is performed at the receiver end, the quantization error spectrum is shaped by the filter $W^{-1}(z)P^{-1}(z)$. When μ is set equal to γ_2 , which is typically the case, the spectrum of the

quantization error is shaped by the filter $1/A(z/\gamma_1)$, with $A(z)$ computed based on the preemphasized speech. Subjective listening showed that
20 this structure of achieving the error shaping by a combination of preemphasis and modified weighting filtering is very efficient for encoding wideband signals, in addition to the advantages of ease of fixed-point algorithmic implementation.

2. The closed-loop pitch analysis is improved to better accommodate wideband signals.

5 The pitch harmonics in AbS coders are usually modeled using a pitch delay T and an associated gain b . The excitation signal $u(n)$ is derived by adding the past excitation at delay T scaled by a gain b to an innovation component from a fixed codebook scaled by a gain g . That is

$$10 \quad u(n) = bv_T(n) + gc_k(n)$$

where $v_T(n)$ is the past excitation at delay T samples. For an improved performance, a fractional delay is usually used. In this case, the past excitation is oversampled to achieve the required higher resolution. In most cases, the pitch predictor can be represented by a filter having a

15 transfer function of the form $1/(1-bz^{-T})$, whose spectrum has a harmonic structure over the entire frequency range, with a harmonic frequency related to $1/T$. In case of wideband signals, this structure is not very efficient since the harmonic frequencies don't cover the entire extended spectrum. The harmonic structure exists only up to a certain frequency,

20 depending on the speech segment. A new method which achieves efficient modeling of the harmonic structure of the speech spectrum uses several forms of low pass filters applied to the past excitation and the one yielding higher prediction gain is selected. When subsample pitch resolution is used, the low pass filters can be incorporated into the

25 interpolation filters used to obtain the higher pitch resolution.

3. At the decoder, the innovative contribution to the excitation is enhanced by filtering it through a preemphasis filter whose coefficients are derived from the level of voicing in speech segment in the subframe.

5 Enhancing the periodicity of the excitation signal improves the quality in case of voiced segments. This was done in the past by filtering the innovation from the fixed codebook through a filter having a transfer function of the form $1/(1-\varepsilon z^{-T})$ where ε is a factor below 0.5 which controls the amount of introduced periodicity. This approach is less efficient in case of wideband signals since it introduces the periodicity over the entire spectrum. A new alternative approach is disclosed whereby the periodicity enhancement is achieved by filtering the innovative signal from the fixed codebook by a filter which emphasizes the high frequencies and reduces the low-frequency contents of the innovation, and whose coefficients are related to the level of periodicity in the signal. In this approach, the innovative contribution is reduced mainly at low frequencies, which enhances the periodicity of the excitation at low frequencies more than high frequencies.

4. A new high-frequency generation procedure is introduced in order to recover the high frequency content of the signal, in case the input signal has been down-sampled.

25 In order to improve the coding efficiency and reduce the algorithmic complexity of the wideband coding algorithm, the input wideband signal is down-sampled from 16 kHz to around 12.8 kHz. This reduces the number of samples in a frame which reduces the processing time, and reduces the signal bandwidth which enables the reduction in bit

rate down to 12 kbit/s while keeping very high quality decoded sound signal. At the decoder, the high frequency contents of the signal needs to be reintroduced to remove the low pass filtering effect from the decoded signal and retrieve the natural sounding quality of wideband signals. A new approach consists of generating the high frequency
5 contents by filling the upper part of the spectrum with a white noise properly scaled in the excitation domain, then converted to the speech domain, preferably but not necessarily by shaping it with the same LP filter used for synthesizing the down-sampled signal.

10 The objects, advantages and other features of the present invention will become more apparent upon reading of the following non restrictive description of a preferred embodiment thereof, given by way of example only with reference to the accompanying drawings.

15

BRIEF DESCRIPTION OF THE DRAWINGS

In the appended drawings:

20

Figure 1 is a schematic block diagram of a preferred embodiment of a wideband encoding device embodying the present invention;

25 ~~Figure 2 is a schematic block diagram of a preferred~~
embodiment of a wideband decoding device embodying the present invention, and comprising a method for high frequency generation; and

Figure 3 is a schematic block diagram of a closed-loop pitch analysis device suitable for wideband signals.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

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The novel techniques disclosed in the present specification may apply to different LP (Linear Prediction)-based coding systems. However, a CELP-type coding system is used in the preferred embodiment for presenting a non limitative illustration of the techniques disclosed herein.

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Figure 1 shows a general block diagram of a CELP-type speech encoding device modified to better accommodate wideband signals.

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The sampled input speech is divided into L -sample blocks called "frames". In each frame, different parameters representing the speech signal in the frame are computed, encoded, and transmitted. LP parameters representing the LP synthesis filter are usually computed once every frame. The frame is further divided into smaller blocks of length N , in which excitation parameters (pitch and innovation) are determined. In the CELP literature, these blocks of length N are called "subframes" and the N -sample signals in a subframe are referred to as N -dimensional vectors. In this preferred embodiment, the length N corresponds to 5-ms while the length L corresponds to 20 ms, which means that a frame contains four subframes ($N=80$ at the sampling rate of 16 kHz and 64 after down-sampling to 12.8 kHz). Various N -dimensional vectors occur in the encoding procedure. A list of the vectors

20

25

which appear in Figures 1 and 2 as well as a list of transmitted parameters are given herein below:

List of the main N -dimensional vectors

5	s Input speech vector (after down-sampling, pre-processing, and preemphasis); s_w Weighted speech vector; s₀ Zero-input response of weighted synthesis filter; x Target vector for pitch search;
10	h Impulse response of the combination of synthesis and weighting filters; v_T Adaptive codebook vector at delay T; y_T Filtered adaptive codebook vector (v_T convolved with h); x' Target vector for pitch search;
15	c_k Innovation codevector at index k (k-th entry from the innovation codebook); c_f Enhanced scaled innovation codevector; u Excitation signal (scaled innovation and pitch codevectors); u' Enhanced excitation;
20	s' Synthesis signal before deemphasis; and s_n Synthesis signal after deemphasis and postprocessing.

List of transmitted parameters

25	STP Short term prediction parameters (defining $A(z)$); T Pitch lag (or adaptive codebook index); b Pitch gain (or adaptive codebook gain);
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- j Index of the low-pass filter used on the pitch codevector;
- k Codevector index (innovation codebook entry); and
- g Innovation codebook gain.

- 5 In this preferred embodiment, the STP parameters are transmitted once per frame and the rest of the parameters are transmitted four times per frame (every subframe).

ENCODING PRINCIPLE

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The sampled speech signal is encoded on a block by block basis by the encoding device of Figure 1 which is broken down into eleven modules numbered from 101 to 111.

- 15 The input speech is processed into the above mentioned L -sample blocks called frames.

- Referring to Figure 1, the input speech signal is down-sampled in a down-sampling module 101. In this preferred embodiment, the signal is down-sampled from 16 kHz down to 12.8 kHz, using techniques well known in the art. Down-sampling increases the coding efficiency, since a smaller bandwidth is encoded. This also reduces the algorithmic complexity since the number of samples in a frame is decreased. The use of down-sampling becomes significant as the bit rate is reduced below 16 kbit/s, although down-sampling is not essential above 16 kbit/s.
- 20
- 25

After down-sampling, the 320-sample frame of 20 ms is reduced to 256-sample frame (down-sampling ratio of 4/5).

The input frame is then passed into the optional pre-processing block 102, which consists of a high pass filter with a 50 Hz cut-off frequency. High-pass filter 102 removes the unwanted sound components below 50 Hz.

5 The down-sampled pre-processed signal is denoted by $s_p(n)$, $n=0,\dots,L-1$, where L is the length of the frame (256 at 12.8 kHz sampling). In preemphasis 103, the signal $s_p(n)$ is preemphasized using a filter having the following transfer function:

10
$$R(z)=1-\mu z^{-1}$$

where μ is a preemphasis factor with a value between 0 and 1 (a typical value is $\mu=0.7$). A higher order filter can also be used.

15 Note that the high-pass filter 102 and preemphasis filter 103 can be interchanged to obtain more efficient fixed-point implementations.

20 The function of the preemphasis filter 103 is to reduce the dynamic range of the input speech signal, which renders it more suitable for fixed-point implementation. Without preemphasis, it is difficult to implement LP analysis in fixed-point using single-precision arithmetic.

25 Preemphasis also plays an important role in achieving a proper overall perceptual weighting of the quantization error, which contributes to an improved sound quality. This will be explained later in more details.

The output of the preemphasis filter 103 is denoted $s(n)$. This signal is used for performing LP analysis, a technique well known in the art. The autocorrelation approach is used, where the signal is first windowed using a Hamming window (usually in the order of 30-40 ms). The autocorrelations are computed from the windowed signal, and
 5 Levinson-Durbin recursion is used to compute the LP parameters, a_i , where $i=1, \dots, p$, and where p is the LP order, which is typically 16 in wideband coding. The parameters a_i are the coefficients of the transfer function of the LP filter:

10
$$A(z) = 1 + \sum_{i=1}^p a_i z^{-i}$$

LP analysis is performed in module 104, which also performs the quantization and interpolation of the LP parameters. The LP coefficients are transformed into another equivalent domain more suitable for quantization
 15 and interpolation purposes. The line spectral pair (LSP) and immitance spectral pair (ISP) domains are two domains in which quantization and interpolation can be efficiently performed. The 16 LP parameters can be quantized in the order of 30 to 50 bits using split or multi-stage quantization, or a combination thereof. The purpose of the interpolation is to enable
 20 updating the LP parameters every subframe while transmitting them once every frame, which improves the coder performance without increasing the bit rate.

25 The following paragraphs will describe the rest of the coding operations performed on a subframe basis. In the following description, the

filter $A(z)$ denotes the unquantized interpolated LP filter in the subframe, and the filter $\hat{A}(z)$ denotes the quantized interpolated LP filter in the subframe.

Perceptual Weighting:

5 In analysis-by-synthesis coders, the optimum pitch and innovation parameters are searched by minimizing the mean squared error between the input speech and synthesized speech in a perceptually weighted domain. This is equivalent to minimizing the error between the weighted input speech and weighted synthesis speech.

10

The weighted signal $s_w(n)$ is computed in a weighted signal generator 105. Traditionally, the weighted signal $s_w(n)$ is computed by a weighting filter having a transfer function $W(z)$ in the form

15
$$W(z) = A(z/\gamma_1)/A(z/\gamma_2) \text{ where } 0 < \gamma_2 < \gamma_1 \leq 1$$

In analysis-by-synthesis (AbS) coders, analysis shows that the quantization error is weighted by a transfer function, $W^{-1}(z)$, which is the inverse of the transfer function of the filter 105. Transfer function $W^{-1}(z)$ exhibits some of the formant structure in the input signal. Thus, the masking property of the human ear is exploited by shaping the error, so that it has more energy in the formant regions, where it will be masked by the strong signal energy present in those regions. The amount of weighting is controlled by the factors γ_1 and γ_2 .

20

The above traditional weighting filter works well with telephone band signals. However, it was found that this weighting filter is not suitable for efficient perceptual weighting when it was applied to wideband signals. It was found that this filter has inherent limitations in modeling the formant structure and the required spectral tilt concurrently. The spectral tilt is more pronounced in wideband signals due to the wide dynamic range between low and high frequencies. The prior art has suggested to add a tilt filter into $W(z)$ in order to control the tilt and formant weighting separately.

A novel solution to this problem, which is part of the present invention, is to introduce the preemphasis filter 103 at the input, compute the LP filter $A(z)$ based on the preemphasized speech $s(n)$, and use a modified filter $W(z)$ by fixing its denominator.

LP analysis is performed in module 104 on the preemphasized signal $s(n)$ to obtain the LP filter $A(z)$. A new perceptual weighting filter 105 with fixed denominator

$$W(z) = A(z/\gamma_1)/(1 - \gamma_2 z^{-1}) \quad \text{where } 0 < \gamma_2 < \gamma_1 \leq 1$$

is used (a higher order can be used at the denominator). This form decouples the formant weighting from the tilt.

Note that because $A(z)$ is computed based on the preemphasized speech signal $s(n)$, the tilt of the filter $1/A(z/\gamma_1)$ is less pronounced compared to the case when $A(z)$ is computed based on the original

speech. Since deemphasis is made at the receiver end using a filter having a transfer function $P^{-1}(z)=1/(1-\mu z^{-1})$, the quantization error spectrum is shaped by a filter having a transfer function $W^{-1}(z)P^{-1}(z)$.

When μ is set equal to γ_2 , which is typically the case, the spectrum of the quantization error is shaped by a filter whose transfer function is

5 $1/A(z/\gamma_1)$, with $A(z)$ computed based on the preemphasized speech. Subjective listening showed that this structure of achieving the error shaping by a combination of preemphasis and modified weighting filtering is very efficient for encoding wideband signals, in addition to the advantages of ease of fixed-point algorithmic implementation.

10

Pitch Analysis:

In order to simplify the pitch analysis, an open-loop pitch lag is first
15 estimated in the open-loop pitch search module 106 using the weighted speech signal $s_w(n)$. Then the closed-loop pitch analysis which is performed in closed-loop pitch search module 107 on a subframe basis is restricted around the open-loop pitch lag which significantly reduces the search complexity of the LTP parameters T and b (pitch lag and pitch
20 gain). Open-loop pitch analysis is usually performed once every 10 ms (two subframes) using techniques well known in the art.

The target signal for LTP (Long Term Prediction) analysis, x , is first
computed. This is usually done by subtracting the zero-input response
25 of a weighted synthesis filter $W(z)/\hat{A}(z)$ (calculated by a zero-input

response generator 108) from the weighted speech signal $s_w(n)$. More specifically, the target vector x is calculated using the following relation:

$$x = s_w - s_0$$

5 where x is the N -dimensional target vector, s_w is the weighted signal vector in the subframe, and s_0 is the zero-input response of the filter $W(z)/\hat{A}(z)$ which is the output of the combined filter $W(z)/\hat{A}(z)$ due to its initial states. s_0 is computed in the zero-input response generator 108.

10 Just a word to mention that alternative, but mathematically equivalent approaches can be used to compute the target vector.

A N -dimensional impulse response vector h of the weighted synthesis filter $W(z)/\hat{A}(z)$ is computed in the impulse response generator 109.

15

The closed-loop pitch or adaptive codebook parameters are computed in the closed-loop pitch search module 107, which uses the target vector x and the impulse response vector h as inputs. Traditionally, the pitch prediction was represented by a pitch filter having the following transfer function:

20

$$1/(1-bz^{-T})$$

where b is the pitch gain and T is the pitch delay or lag. In this case, the pitch contribution to the excitation signal $u(n)$ is given by $bu(n-T)$, where the total excitation is given by

25

$$u(n) = bu(n-T) + gc_k(n)$$

with g being the innovative codebook gain and $c_k(n)$ the innovation codevector at index k .

- 5 This representation has limitations if the delay T is shorter than the subframe length N . In another view point, the pitch contribution can be seen as an adaptive codebook containing the past excitation signal. Generally, each vector in the adaptive codebook is a shift-by-one version of the previous vector (discarding one sample and adding a new sample). For
10 delays $T > N$, the adaptive codebook is equivalent to the filter structure, and a codevector $v_T(n)$ is given by

$$v_T(n) = u(n-T), \quad n=0, \dots, N-1.$$

- 15 For delays shorter than T , a codevector is built by repeating the available samples from the past excitation until the codevector is completed (this is not equivalent to the filter structure).

- 20 In recent coders, a higher pitch resolution is used which significantly improves the quality of voiced sound segments. This is achieved by oversampling the past excitation signal using polyphase interpolation filters. In this case, the codevector $v_T(n)$ may correspond to an interpolated version of the past excitation, with T being a non-integer delay (e.g. 50.25).

- 25 The pitch search consists of finding the best delay T and gain b that minimize the mean squared weighted error between the target vector x and the scaled filtered past excitation

$$E = \|x - by_T\|^2$$

where y_T is the filtered adaptive codevector at delay T :

$$y_T(n) = v_T(n) * h(n) = \sum_{i=0}^n v_T(i) h(n-i), \quad n=0, \dots, N-1.$$

5

It can be shown that the error E is minimized by maximizing the criterion

$$C = \frac{x' y_T}{\sqrt{y_T' y_T}}$$

10 where t denotes vector transpose.

In the preferred embodiment of the present invention, a 1/3 subsample pitch resolution is used, and the pitch search is composed of three stages.

15

In the first stage, an open-loop delay is estimated in open-loop pitch search module 106. In the second stage, the search criterion C is searched in the closed-loop pitch search module 107 for integer delays around the estimated open-loop delay (usually ± 5), which significantly simplifies the
 20 search procedure. A simple procedure is used for updating the filtered codevector y_T without the need to compute the convolution for every delay. Once an optimum integer delay is found, the fractions around the integer delay are tested in the third stage of the search (module 107).

When the pitch predictor is represented by a filter of the form $1/(1-bz^{-T})$, which is a valid assumption for delays $T > N$, the spectrum of the pitch filter exhibits a harmonic structure over the entire frequency range, with a harmonic frequency related to $1/T$. In case of wideband signals, this structure is not very efficient since the harmonic structure in
5 wideband signals does not cover the entire extended spectrum. The harmonic structure exists only up to a certain frequency, depending on the speech segment. Thus, in order to achieve efficient representation of the pitch contribution in voiced segments of wideband speech, the pitch predictor need to have the flexibility of varying the amount of periodicity
10 over the wideband spectrum.

A new method which achieves efficient modeling of the harmonic structure of the speech spectrum is disclosed in the present specification, whereby several forms of low pass filters are applied to the past excitation
15 and the one with higher prediction gain is selected.

When subsample pitch resolution is used, the low pass filters can be incorporated into the interpolation filters used to obtain the higher pitch resolution. In this case, the third stage of the pitch search, in which the
20 fractions around the chosen integer delay are tested, is repeated for the several interpolation filters having different low-pass characteristics and the fraction and filter index which maximize the search criterion C are selected.

25 A simpler approach, is to complete the search in the three stages described above, to determine the optimum fractional delay using only one interpolation filter with certain frequency response, and select the

optimum low-pass filter shape at the end by applying the different pre-determined low-pass filters to the chosen adaptive codevector v_T and select the low-pass filter which minimizes the pitch prediction error.

Figure 3 shows a schematic block diagram of a preferred embodiment of the proposed approach.

In module 303, the past excitation codevector is memorized. Module 301 is responsive to the target vector x and to the past excitation codevector from memory module 303 to conduct a pitch codebook search minimizing the above-defined search criterion C . From the result of the search conducted in module 301, module 302 generates the optimum codevector v_T .

Suppose that K filter characteristics are used (they could be low-pass or band-pass). Once the optimum codevector v_T is determined, K filtered versions of v_T are computed using the K different frequency shaping filters such as 305⁰, where $j=1, \dots, K$. These filtered versions are denoted

$v_f^{(j)}, j=1, \dots, K$. The different vectors $v_f^{(j)}$ are convolved in modules 304⁰,

where $j=1, \dots, K$, with the impulse response h to obtain the vectors $y^{(j)}, j=1, \dots, K$. The selected frequency shaping filter 305⁰ is the one which minimizes the mean squared pitch prediction error

$$e^{(j)} = \|x - b^{(j)} y^{(j)}\|^2, \quad j=1, \dots, K$$

To calculate the mean squared pitch prediction error for each value of $y^{(j)}$, the

value $y^{(j)}$ is multiplied by the gain b by means of an amplifier 307^(j) and the value $b^{(j)}y^{(j)}$ is subtracted from the target vector x by means of subtractors 308^(j).

5 The gain $b^{(j)}$ associated with the frequency shaping filter at index j ,
is given by

$$b^{(j)} = x' y^{(j)} / \|y^{(j)}\|^2.$$

10 In the same manner, optimum codevector v_T is convolved with the impulse response h to obtain the vectors y . To calculate the mean squared pitch prediction error for y , the value y is multiplied by the gain b by means of an amplifier 307^(j) and the value by is subtracted from the target vector x by means of subtractors 308. The gain b is given by

15

$$b = x' y / \|y\|^2$$

In module 309, the parameters b , T , and j are chosen based on v_T or $v_T^{(j)}$ which minimizes the mean squared pitch prediction error e .

20 The pitch codebook index T is encoded and transmitted. The pitch gain b is quantized and transmitted. With this new approach, extra information is needed to encode the index j of the selected frequency shaping filter. If two filters are used, then one bit is needed to represent this information.

25

Innovative codebook search:

Once the pitch, or LTP (Long Term Prediction) parameters b , T , and j are determined, we proceed by searching for the optimum innovative excitation by means of module 110 of Figure 1. First, the target vector $\mathbf{x}$ is
 5 updated by subtracting the LTP contribution:

$$\mathbf{x}' = \mathbf{x} - b\mathbf{y}_T$$

where b is the pitch gain and $\mathbf{y}_T$ is the filtered adaptive codebook vector (the
 10 past excitation at delay T filtered with the selected low pass filter and convolved with the impulse response $\mathbf{h}$ as described with reference to Figure 3).

The search procedure in CELP is performed by finding the optimum
 15 excitation codevector $\mathbf{c}_k$ and gain g which minimize the mean-squared error between the target vector and the scaled filtered codevector

$$E = \|\mathbf{x}' - g\mathbf{H}\mathbf{c}_k\|^2$$

20 where $\mathbf{H}$ is a lower triangular convolution matrix derived from the impulse response vector $\mathbf{h}$.

In the preferred embodiment of the present invention, the innovative codebook search is performed in module 110 by means of an algebraic
 25 codebook as described in US patent numbers 5,444,816 (Adoul et al.) issued on August 22, 1995; 5,699,482 granted to Adoul et al., on December

17, 1997; 5,754,976 granted to Adoul et al., on May 19, 1998; and 5,701,392 (Adoul et al.) dated December 23, 1997.

Once the optimum codevector and its gain are chosen by module 110, the codebook index k and gain g are encoded and transmitted.

5

Referring to Figure 1, the parameters b , T , j , $\hat{A}(z)$, k and g are multiplexed through a multiplexer 112 before being encoded and transmitted

10 **Memory update:**

In module 111 (Figure 1), the states of the weighted synthesis filter are updated by filtering the excitation signal $u = g\mathbf{c}_k + b\mathbf{v}_T$ through the weighted synthesis filter. At the end of this filtering, the states of the filter are memorized and used in the next subframe as initial states for computing the zero-input response in generator module 108.

15

Similar to the target vector, other alternative, but mathematically equivalent, approaches can be used to update the filter states.

20

DECODING PRINCIPLE

The speech decoding device of Figure 2 illustrates the various steps carried out between the digital input 222 (input to the demultiplexer 217) and the output sampled speech 223 (output of the adder 221).

25

The demultiplexer 217 extracts the synthesis model parameters from the binary information received from a digital input channel. From each received binary frame, the extracted parameters are:

- the short-term prediction parameters STP (once per frame);
- the long-term prediction (LTP) parameters T , b , and j (for each subframe); and
- the innovation codebook index k and gain g (for each subframe).

The current speech signal is synthesized based on these parameters as will be explained hereinbelow.

The innovative excitation generator 218 is responsive to the index k to produce the innovation codevector $\mathbf{c}_k$, which is scaled by the decoded gain factor g through an amplifier 224. In the preferred embodiment, an algebraic codebook as described in the above mentioned US patent numbers 5,444,816; 5,699,482; 5,754,976; and 5,701,392 is used to represent the innovative excitation.

The generated scaled codevector at the output of the amplifier 224 is processed through a frequency-dependent pitch enhancer 205.

Enhancing the periodicity of the excitation signal improves the quality in case of voiced segments. This was done in the past by filtering the innovation from the fixed codebook through a filter in the form $1/(1-\varepsilon z^{-T})$ where ε is a factor below 0.5 which controls the amount of

introduced periodicity. This approach is less efficient in case of wideband signals since it introduces the periodicity over the entire spectrum. A new alternative approach, which is part of the present invention, is disclosed whereby the periodicity enhancement is achieved by filtering the innovative signal from the fixed codebook by a filter $F(z)$ whose frequency response emphasizes the higher frequencies more than lower frequencies. The coefficients of $F(z)$ are related to the amount of periodicity in the signal. An efficient way to derive the filter coefficients is to relate them to the amount of pitch contribution to the total excitation. This results in a frequency response depending on the subframe periodicity, where higher frequencies are more strongly emphasized (stronger overall slope) for higher pitch gains. This filter has the effect of lowering the energy of the innovative excitation at low frequencies when the signal is more periodic, which enhances the periodicity of the excitation at lower frequencies more than higher frequencies. Suggested forms of this filter are

$$(1) \quad F(z) = 1 - \alpha^{-1} \quad \text{or} \quad (2) \quad F(z) = -\alpha + 1 - \alpha^{-1}$$

where σ or α are factors derived from the level of periodicity of the signal. The second 3-tape form of $F(z)$ is used in this preferred embodiment. The factor α is computed in the voicing factor generator 204 as follows: The ratio of pitch contribution to the total excitation is first computed by

$$R_p = \frac{b^2 \mathbf{v}_T' \mathbf{v}_T}{\mathbf{u}' \mathbf{u}} = \frac{b^2 \sum_{n=0}^{W-1} v_T^2(n)}{\sum_{n=0}^{W-1} u^2(n)}$$

where $\mathbf{v}_T$ is the pitch codebook vector, b is the pitch gain, and $\mathbf{u}$ is the excitation vector given at the output of the adder 219 by

$$\mathbf{u} = b\mathbf{v}_T + g\mathbf{c}_k$$

5

Just a word to mention that the term $b\mathbf{v}_T$ is produced by the pitch codebook 201 in response to the pitch lag T and the past value of $\mathbf{u}$ stored in memory 203. The adaptive codevector from the pitch codebook 201 is then processed through a low-pass filter whose cut-off frequency
10 is adjusted by means of the index j from the demultiplexer 217. The resulting codevector $\mathbf{v}_T$ is then multiplied by the gain g from the demultiplexer 217 through an amplifier 226 to obtain the signal $b\mathbf{v}_T$.

The factor α is given by

$$15 \quad \alpha = qR_p \quad \text{bounded by } \alpha < q$$

where q is a factor which controls the amount of enhancement (q is set to 0.25 in this preferred embodiment).

The enhanced signal $\mathbf{c}_f$ is computed by filtering the scaled
20 innovative vector $g\mathbf{c}_k$ through the enhancing filter $F(z)$.

The enhanced excitation signal $\mathbf{u}'$ is computed by the adder 220
as

$$25 \quad \mathbf{u}' = b\mathbf{v}_T + \mathbf{c}_f$$

Note that this process is not performed at the encoder. Thus, it is essential to update the content of the adaptive codebook using the excitation without enhancement to keep synchronism between the encoder and decoder. Therefore, the excitation signal u is used to update the memory of the adaptive codebook and the enhanced excitation signal u' is used at the input of the LP synthesis filter 206.

The synthesized signal s' is computed by filtering the enhanced excitation signal u' through the LP synthesis filter 206 which has the form $1/\hat{A}(z)$, where $\hat{A}(z)$ is the interpolated LP filter in the current subframe. As can be seen in Figure 2, the LP coefficients 225 from the demultiplexer 217 are supplied to the LP filter 206 to adjust the parameters of the LP filter 206 accordingly. The deemphasis filter 207 is the inverse of the preemphasis filter 103 of Figure 1. The transfer function of the preemphasis filter 108 is given by

$$D(z) = 1/(1 - \mu z^{-1})$$

The vector s' is filtered through the deemphasis filter $D(z)$ (module 207) to obtain the vector s_d , which is passed through the postprocessing module 208 comprising a high-pass filter to remove the unwanted frequencies below 50 Hz.

The over-sampling module 209 conducts the inverse process of the down-sampling module 101 of Figure 1. In this preferred embodiment, oversampling converts from the 12.8 kHz sampling rate to the original 16 kHz sampling rate, using techniques well known in the art. The oversampled synthesis signal is denoted $\hat{s}$.

The synthesis signal does not contain the higher frequency components which were lost by the downsampling process (module 101 of Figure 1) at the encoder. This gives a low-pass perception of the synthesis speech. To restore the full band of the original signal, a high frequency generation procedure is disclosed. This procedure is performed in modules
5 212 through 216 of Figure 2.

In this new approach, the high frequency contents are generated by filling the upper part of the spectrum with a white noise properly scaled in the excitation domain, then converted to the speech domain, preferably by
10 shaping it with the same LP filter used for synthesizing the down-sampled signal.

The high frequency generation procedure, which is part of the present invention, is detailed hereinbelow.
15

The random noise generator 213 generates a white noise sequence w' with a flat spectrum over the entire frequency bandwidth, using techniques well known in the art. The generated sequence is of length N' which is the subframe length in the original domain. Note that N is the
20 subframe length in the down-sampled domain. In this preferred embodiment, $N=64$ and $N'=80$ which correspond to 5 ms.

The white noise sequence is properly scaled in the gain adjusting module 214. Gain adjustment comprises the following steps. First, the
25 energy of the generated noise sequence is set equal to the energy of the enhanced excitation signal u' computed by an energy computing module 210, and the resulting scaled noise sequence w is given by

$$w(n) = w'(n) \sqrt{\frac{\sum_{n=0}^{N'-1} u^2(n)}{\sum_{n=0}^{N'-1} w'^2(n)}}, \quad n=0, \dots, N'-1$$

5 The second step in the gain scaling is to take into account the voicing of the synthesized signal at the output of generator 204 so as to reduce the energy of the generated noise proportional to the voicing. In this preferred embodiment, this is implemented by measuring the tilt of the synthesis signal through a spectral tilt calculator 212 and reducing the energy accordingly. When the tilt is very strong, which corresponds to voiced segments, the noise energy is further reduced. The tilt factor is computed in module 212 as the first correlation coefficient of the synthesis signal s_h and it is given by

10

$$tilt = \frac{\sum_{n=1}^{N-1} s_h(n) s_h(n-1)}{\sum_{n=0}^{N-1} s_h^2(n)}, \quad \text{bounded by } tilt \geq 0 \text{ and } tilt \geq r_v$$

r_v is given by

15 $r_v = (E_v - E_c) / (E_v + E_c)$ where E_v is the energy of the scaled pitch codevector and E_c is the energy of the scaled innovative codevector. r_v is mostly less than $tilt$ but this bound was introduced as a precaution against high frequency tones where the tilt value is high and the value of r_v is small. So this bound reduces the noise energy for such tonal signals.

20 The tilt value is 0 in case of flat spectrum and 1 in case of strongly voiced signals. The scaling factor derived from the tilt is given by

$$g_i = 10^{-0.6 \text{ tilt}}$$

When the tilt is close to zero, the scaling factor is close to 1, which does not result in energy reduction. When the tilt value is 1, the scaling factor results in a reduction of 12 dB in the energy of the generated noise.

5

Once the noise is properly scaled, it is brought into the speech domain using the spectral shaper 215. In the preferred embodiment, this is achieved by filtering the noise through a bandwidth expanded version of the same LP synthesis filter used in the down-sampled domain ($1/\hat{A}(z/0.8)$).

10

The filtered scaled noise sequence is then band-pass filtered to the required frequency range to be restored using the band-pass filter 216. In the preferred embodiment, the band-pass filter 216 restricts the noise sequence to the frequency range 5.6-7.2 kHz. The resulting band-pass noise sequence z is added to the oversampled synthesized speech signal $\hat{s}$ to obtain the final reconstructed sound signal s_{out} on the output 223.

15

Although the present invention has been described hereinabove by way of a preferred embodiment thereof, this embodiment can be modified at will, within the scope of the appended claims, without departing from the spirit and nature of the subject invention.

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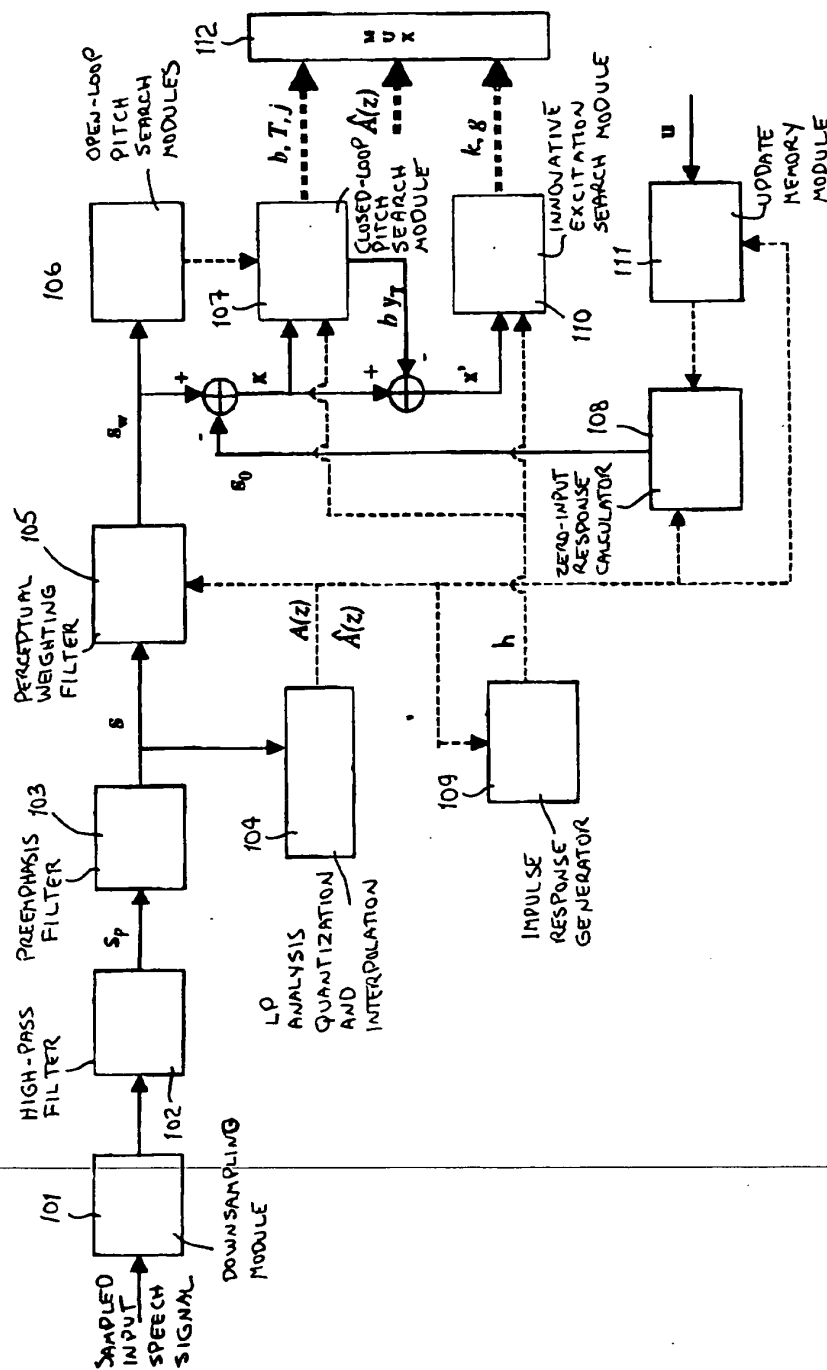
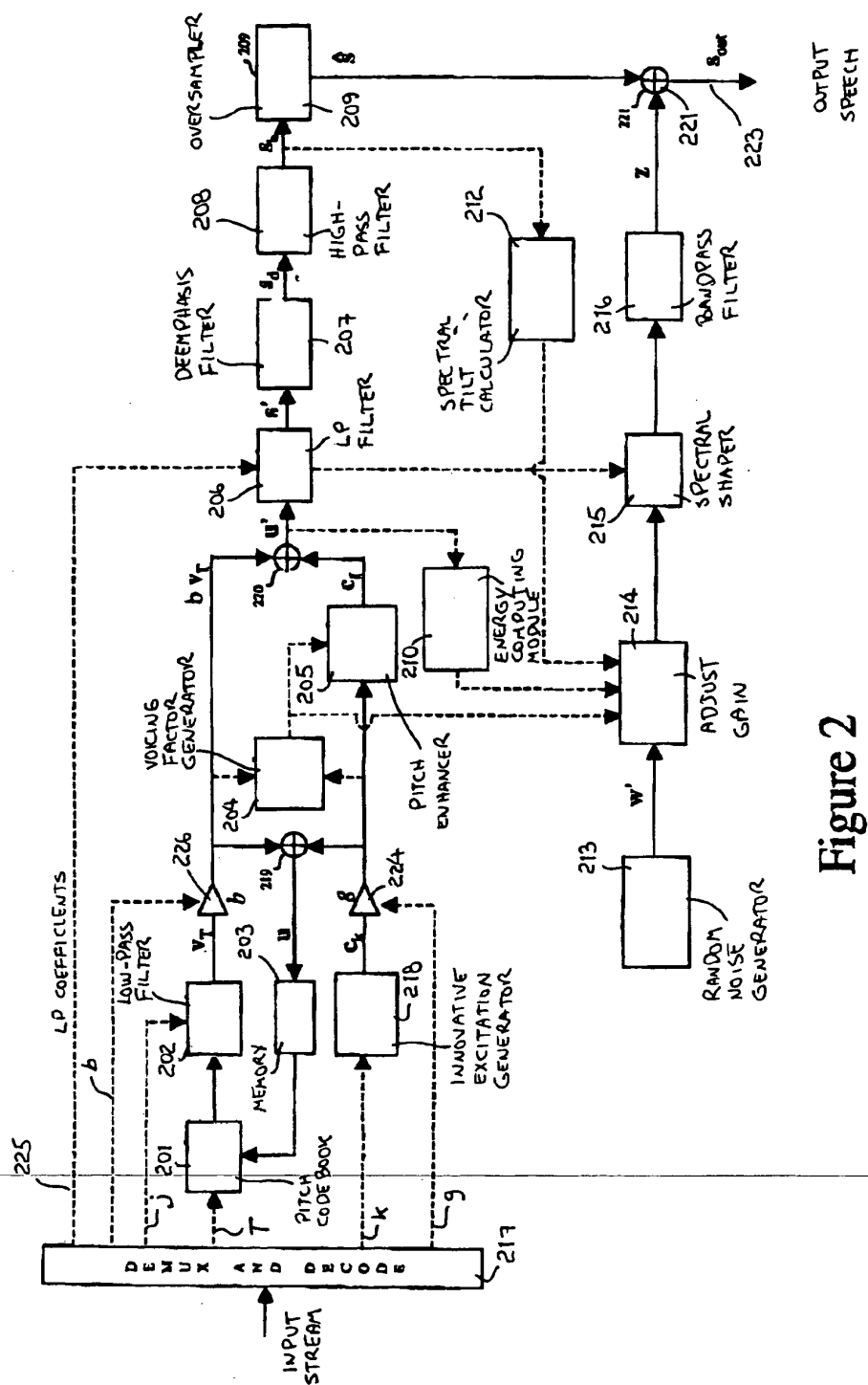


Figure 1



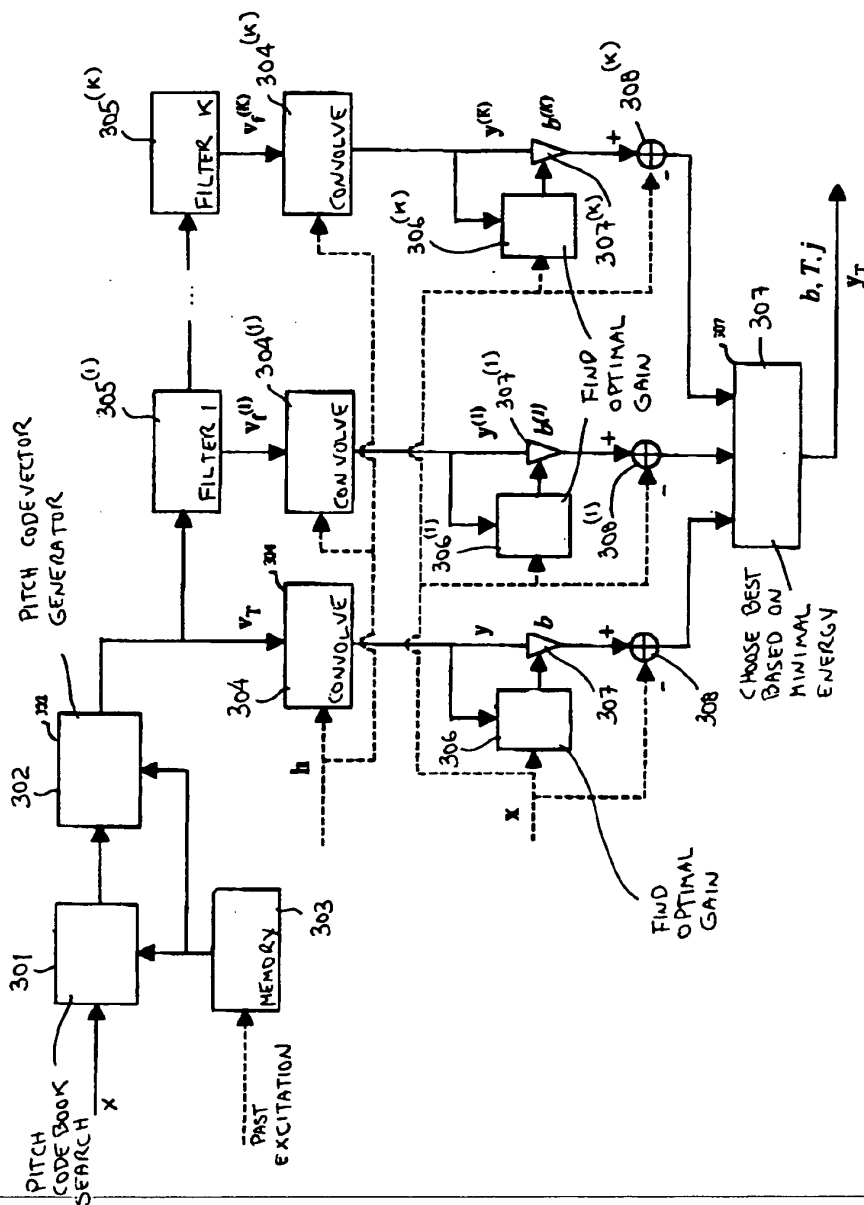


Figure 3

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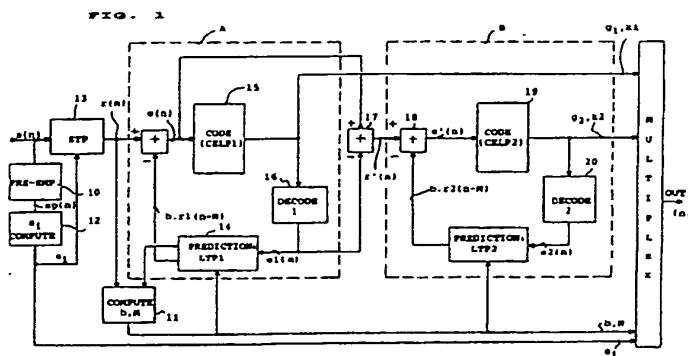
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(54) Multi-rate voice encoding method and device.

57) The voice signal $s(n)$ is filtered through a short-term predictive filter (13) tuned with PARCOR derived coefficients computed over a pre-emphasized $s(n)$, said filter (13) providing a short-term residual $r(n)$. Said $r(n)$ signal is then processed through a first Code-Excited/Long-Term Predictive coder providing first couples of table address and gain data (k_1, g_1)'s. An error signal $r'(n)$ is then derived by subtracting coded/decoded data from uncoded data. Then said error signal is processed through a second Code-Excited/ Long-Term Predictive coder providing second couples of data (k_2, g_2)'s. Full rate coding is achieved by multiplexing both couples (k_1, g_1)'s and (k_2, g_2)'s into a multi-rate frame; while switching to a lower rate is achieved through a mere delation of (g_2, k_2)'s from the full rate frame.



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MULTI-RATE VOICE ENCODING METHOD AND DEVICE

Technical Field of the Invention

This invention deals with voice coding techniques and more particularly with a method and means for
 5 multi-rate voice coding.

Background of the Invention

10 Digital networks are currently used to transmit, and/or store where convenient, digitally encoded voice signals. For that purpose, each voice signal to be considered is, originally, sampled and each sample digitally encoded into binary bits. In theory, at least, the higher the number of bits used to code each sample the better the coding, that is the closest the voice signal would be when decoded before being
 15 provided to the end user. Unfortunately, for the network to be efficient from an economical stand point, the traffic or in other words the number of connected users acceptable without network congestion needs be maximized. This is one of the reasons why methods have been provided for lowering the voice coding bit rates while keeping the coding distortion (noise) at acceptable levels, rather than dropping users when traffic increases over a network. It looks reasonable to improve the voice coding quality when the traffic
 20 permits it and if needed lower said quality to a predetermined acceptable level under high traffic conditions. This switching from one quality (one bit rate) to another, should be made as simple and quick as possible at any node within the network. For that purpose, multirate coders should provide frames with embedded bit streams whereby switching from one predetermined bit rate to a lower predetermined rate would simply require dropping a predetermined portion of the frame.

Summary of the Invention

30 One object of this invention is to provide means for multi-rate coding a voice signal using Code-Excited encoding techniques.

The voice signal is short-term filtered to derive a short-term residual therefrom, said short-term residual is submitted to a first Long-Term Predictive Code-Excited coding operation, then decoded and subtracted from the Code-Excited coding input to derive an Error signal, which Error signal is in turn Long-Term
 35 Predictive Code-Excited coded. Multi-rate frame involves both Long-Term Predictive Code-Excited coding.

More particularly, the present invention processes by short-term filtering the original voice signal to derive a voice originating short-term residual signal; submitting said short-term residual to a first Code-Excited (CE) coding operation including : subtracting from said short-term residual a first predicted residual
 40 signal to derive a first long-term residual signal, coding said long term residual into a gain g_1 and an address k_1 ; subtracting said first reconstructed residual (after decoding) from the first long-term residual to derive a first Error signal therefrom; submitting said first Error signal to subsequent Code-Excited long-term prediction coding into g_2 and k_2 ; and aggregating (g_1 , k_1) and (g_2 , k_2) into a same multi-rate coded frame, whereby switching to a lower rate coded frame would be achieved through dropping (g_2 , k_2).

Obviously, the above principles may be extended to a higher number of rates by extending it to third,
 45 fourth, etc. ... Code-Excited coding.

Further objects, characteristics and advantages of the present invention will be explained in more details in the following, with reference to the enclosed drawings, which represent a preferred embodiment.

The foregoing and other objects, features and advantages of the invention will thereof be made apparent from the following more particular description of a preferred embodiment of the invention as
 50 illustrated in the accompanying drawings.

Brief Description of the Drawings

- Figure 1 : is a block diagram of a coder according to the invention.
- Figure 2 : is a flow chart for the operations involved in devices 10, 12 and 13 of figure 1.
- Figure 3 : is a flow chart for Code-Excited coding operations.
- Figure 4 : is a block diagram for implementing the device 14 of figure 1.
- Figure 5 : is a flow chart of the process of the invention as applied to device of figure 1.
- Figure 6 : is a flow chart for the decoder to be used with the invention.
- Figure 7 : is a block diagram of said decoder.
- Figure 8 : is a block diagram for the coder according to the invention, applied to base-band coding.

Description of Preferred Embodiment

Represented in figure 1 is a simplified block diagram of a bi-rate coder, which, as already mentioned, might be extended to a higher number of rates.

The voice signal limited to the telephone bandwidth (300 Hz-3300 Hz), sampled at 8 KHz and digitally PCM encoded with 12 bits per sample in a conventional Analog to Digital Converter (not shown) provides samples $s(n)$. These samples are first pre-emphasized in a device (10) and then processed in a device (12) to generate sets of partial autocorrelation derived coefficients (PARCOR derived) a_i 's. Said a_i coefficients are used to tune a short term predictive filter (STP) (13) filtering $s(n)$ and providing a short-term residual signal $r(n)$. Said short-term residual is coded into a first Code-Excited long-term prediction coder (A). To that end, it is processed to derive therefrom a first long-term residual $e(n)$ by subtracting from $r(n)$, a predicted first residual signal corresponding to the synthesized (reconstructed) first residual delayed by a predetermined delay M (equal to a multiple of the voice pitch period) and multiplied by a gain factor $b \cdot r1(n-M)$ using as first long-term predictor.

It should be noted that for the purpose of this invention block coding techniques are used over $r(n)$ blocks of samples, 160 samples long. Parameters b and M are evaluated every 80 samples. The flow of residual signal samples $e(n)$ is subdivided into blocks of L consecutive samples and each of said blocks is then processed into a first Code-Excited coder (CELP1) (15) where K sequences of L samples are made available as normalized codewords. Coding $e(n)$ involves then selecting the codeword best matching the considered $e(n)$ sequence in mean squared error criteria consideration and replacing $e(n)$ by a codeword reference number $k1$. Assuming the pre-stored codewords be normalized, then a first gain coefficient $g1$ should also be determined and tested.

Once $k1$ is determined, a first reconstructed residual signal $e1(n) = g1 \cdot CB(k1)$ generated in a first decoder (DECODE1) (16) is fed into said long-term predictor (14).

Said reconstructed residual is also subtracted from $e(n)$ in a device (17) providing an error signal $r'(n)$.

The error signal $r'(n)$ is then fed into a second Code-Excited/Long-Term Prediction coder similar to the one described above. Said second coder includes a subtractor (18) fed with the error signal $r'(n)$ and providing an error residual signal $e'(n)$ addressing a second Code-Excited coder CELP2 (19). Said device (19) codes $e'(n)$ into a gain factor $g2$ and a codeword address $k2$. Said coder is also made to feed the codeword $CB(k2)$ and gain $g2$ into a decoder (20) providing a decoded error signal $e2(n) = g2 \cdot CB(k2)$.

Said signal $e2(n)$ is also fed into a second Long-Term Predictor (LTP2) similar to LTP1 and the output of which is subtracted from $r'(n)$ in device (18).

Finally a full rate frame is generated by multiplexing the a_i 's b 's, M 's, $(g1, k1)$'s and $(g2, k2)$'s data into a multirate (bi-rate) frame.

As already mentioned, the process may easily be further extended to higher rates by serially inserting additional Code-Excited/Long-Term Predictive coders such as A or B.

Represented in figure 2 is a flow chart showing the detailed operations involved in both pre-emphasis and PARCOR related computations. Each block of 160 signal samples $s(n)$ is first processed to derive two first values of the signal auto-correlation function :

$$R1 = \sum_{j=1}^{160} s(j) \cdot s(j)$$

$$R2 = \sum_{j=2}^{160} s(j-1) \cdot s(j)$$

The pre-emphasis coefficient R is then computed

$$R = R1/R2$$

and the original set of 160 samples s(n) are converted into a pre-emphasized set sp(n)

$$sp(n) = s(n) - R \cdot s(n-1)$$

The pre-emphasized a_i parameters are derived by a step-up procedure from so-called PARCOR coefficients K_i in turn derived from the pre-emphasized signal sp(n) using a conventional Leroux-Guegen method. The eight a_i or PARCOR K_i coefficients may be coded with 28 bits using the Un/Yang algorithm. For reference to these methods and algorithm, one may refer to :

- J. Leroux and C. Guegen : "A fixed point computation of partial correlation coefficients" IEEE Transactions on ASSP pp 257-259, June 1977;
- C.K. Un and S.C. Yang "Piecewise linear quantization of LPC reflexion coefficients" Proc. Int. Conf. on QSSP Hartford, May 1977.
- L.D. Markel and A.H. Gray : "Linear prediction of speech" Springer Verlag 1976, Step-up procedure pp 94-95.
- European Patent 2998 (US patent 4216354) assigned to this assignee.

The short term filter (13) derives the short-term residual signal samples :

$$r(n) = s(n) - \sum_{i=1}^8 a_i \cdot s(n-i)$$

Several methods are available for computing the long-term factors b and M values. One may for instance refer to B.S. Atal "Predictive Coding of Speech at low Bit Rate" published in IEEE Trans on Communication, Vol. COM-30, April 1982, or to B.S. Atal and M.R. Schroeder, "Adaptive prediction coding of speech signals", Bell System Technical Journal; Vol 49, 1970.

Generally speaking, M is a pitch value or an harmonic of it and methods for computing it are known to a man skilled in the art.

A very efficient method was also described in a copending European application (cf FR987004) to the same assignee.

According to said application :

$$b = \frac{\sum_{n=1}^{80} r(n) \cdot r1(n-M)}{\sum_{n=1}^{80} [r1(n-M)]^2}$$

with b and M being determined twice over each block of 160 samples, using 80 samples and their 80 predecessors.

The M value, i.e. a pitch related value, is therein computed based on a two-step process. A first step enabling a rough determination of a coarse pitch related M value, followed by a second (fine) M adjustment

using auto-correlation methods over a limited number of values.

1. First step :

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Rough determination is based on use of non-linear techniques involving variable threshold and zero crossing detections more particularly this first step includes :

- initializing the variable M by forcing it to zero or a predefined value L or to previous fine M;
- loading a block vector of 160 samples including 80 samples of current sub-block, and the 80 previous samples;
- detecting the positive (Vmax) and negative (Vmin) peaks within said 160 samples;
- computing thresholds :
positive threshold $Th^+ = \alpha \cdot Vmax$
negative threshold $Th^- = \alpha \cdot Vmin$
- alpha being an empirically selected value (e.g. $\alpha = 0.5$)
- setting a new vector X(n) representing the current sub-block according to :

$$\begin{aligned} X(n) &= 1 \text{ if } r(n) \geq Th^+ \\ X(n) &= -1 \text{ if } r(n) \leq Th^- \\ X(n) &= 0 \text{ otherwise} \end{aligned}$$

- This new vector containing only -1, 0 or 1 values will be designated as "cleaned vector";
- detecting significant zero crossings (i.e. sign transitions) between two values of the cleaned vector i.e. zero crossing close to each other;
 - computing M' values representing the number of r(n) sample intervals between consecutive detected zero crossings;
 - comparing M' to the previous rough M by computing $\Delta M = |M' - M|$ and dropping any M' value whose ΔM is larger than a predetermined value D (e.g. $D = 5$);
 - computing the coarse M value as the mean value of M' values not dropped.

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2. Second step :

Fine M determination is based on the use of autocorrelation methods operated only over samples taken around the samples located in the neighborhood of the pitched pulses.

Second step includes:

- Initializing the M value either as being equal to the rough (coarse) M value just computed assuming it is different from zero, otherwise taking M equal to the previous measured fine M;
- locating the autocorrelation zone of the cleaned vector, i.e. a predetermined number of samples about the rough pitch;
- computing a set of R(k') values derived from :

$$R(k') = \sum_{n=1}^{80} r(n) \cdot r1(n-k)$$

with k' being the cleaned vector sample index varying from a lower limit Mmin to the upper limit Mmax of the selected autocorrelation zone, with limits of the autocorrelation zone $Mmin = L$, $Mmax = 120$ for example.

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Once b and M are computed, they are used to tune the inverse Long-Term Predictor (14) as will be described further. The output of the device (14) i.e. a predicted first long-term residual subtracted to r(n) provides first long-term residual signal e(n). Said e(n) is in turn, coded into a coefficient k1 and a gain factor g1. The coefficient k1 represents the address of a codeword CB(k1) pre-stored into a table located in the device (CELP1) (15). The codeword and gain factor selection is based on a mean squared error criteria consideration; i.e. by looking for the k table address providing a minimal E, with :

55

$$E = [e(n) - g1 \cdot CB(k,n)]^T \cdot [e(n) - g1 \cdot CB(k,n)] \quad (1)$$

wherein :

T : means mathematical transposition operation. CB(k,n) = represents the codeword located at the address k within the coder 15 of figure 1.

In other words, E is a scalar product of two L components vectors, wherein L is the number of samples of each codeword CB.

The optimal scale factor G(k) [g1 in (1)] that minimizes E is determined by setting :

$$\frac{dE}{dG} = 0$$

$$\text{and } G(k) = \frac{e(n)^T \cdot CB(k,n)}{\|CB(k,n)\|^2}$$

The denominator of equation G(k) is a normalizing factor which could be avoided by pre-normalizing the codewords within the pre-stored table.

The expression (1) can be reduced to :

$$E = \|e(n)\|^2 - \frac{[e(n)^T \cdot CB(k,n)]^2}{\|CB(k,n)\|^2} \quad (2)$$

and the optimum codeword is obtained by finding k maximizing the last term of equation (2).

Let CB2(k) represent $\|CB(k,n)\|^2$ and, SP(k) be the scalar product $e^T(n) \cdot CB(k,n)$,

Then one has first to find k providing a term

$$\frac{[SP(k)]^2}{CB2(k)}$$

maximum, and then determine the G(k) value from

$$G = \frac{SP(k)}{CB2(k)}$$

The above statements could be differently expressed as follows :

Let {en} with n = 1, 2, ..., L represent the sequence of e(n) samples to be encoded. And let {Y_n^k} with n = 1, 2, ..., L and k = 1, 2, ..., K, where K = 2^{cbt}, represent a table containing K codewords of L samples each.

The CELP encoding would lead to :

- computing correlation terms :

$$E_k = \sum_{n=1}^L e_n \cdot Y_n^k$$

for k = 1, ..., K

- selecting the optimum value of k leading to $E_{kopt} = \text{Max} (E_k)$

$k = 1, \dots, K$

- converting the $e(n)$ sequence into a block of $\text{cbit} = \log_2 K$ bits, plus the $G(k)$ encoding bits.

The algorithm for performing the above operations is represented in figure 3.

5 First two index counters i and j are set to $i = 1$ and $j = 1$. The table is sequentially scanned. A codeword $CB(1,n)$ is read out of the table.

A first scalar product is computed

$$10 \quad SP(1) = \sum_{n=1}^L CB(1,n) \cdot e(n) \quad (3)$$

This value is squared into $SP2(1)$ and divided by a squared value of the corresponding codeword [i.e. $CB2(1)$]. i is then incremented by one and the above operations are repeated until $i = K$, with K being the number of codewords in the code-book. The optimal codeword $CB(k)$, which provides the maximum

$$20 \quad \frac{SP2(k)}{CB2(k)}$$

within the sequence

$$25 \quad \frac{SP2(i)}{CB2(i)} \quad \text{for } i = 1, \dots, K$$

30 is then selected. This operation enables detecting the table reference number k.

Once k is selected, then the gain factor computed using :

$$35 \quad G = \frac{SP(k)}{CB2(k)}$$

Assuming the number of samples within the sequence $e(n)$ is selected to be a multiple of L, then said sequence $e(n)$ is subdivided into JL windows each L samples long, then j is incremented by 1 and the above process is repeated until $j = JL$.

40 Computations may be simplified and the coder complexity reduced by normalizing the codebook in order to set each codeword energy to the unit value. In other words, the L component vector amplitude is normalized to one

$CB2(i) = 1$ for $i = 1, \dots, K$

45 In that case, the expression determining the best codeword k is simplified (all the denominators involved in the algorithm are equal to the unit value). The scale factor $G(k)$ is changed whereas the reference number k for the optimal sequence is not modified.

This method would require a memory fairly large to store the table. For instance said size $K \times L$ may be 50 of the order of 40 kilobits for $K = 256$ and $L = 20$.

A different approach is recommended here. Upon initialisation of the system, a first block of $L + K$ samples of residual signal, e.g. $e(n)$ would be stored into a table. Then each subsequent L-word long sequence $e(n)$ is correlated with the $(L + K)$ samples long table sequence by shifting the $\{e(n)\}$ sequence from one sample position of the next, over the table.

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This method enables reducing the memory size required for the table, down to 2 kilobits for $K = 256$, $L = 20$ or even lower.

$$e!(n) = q1 . CB(k1)$$

As represented in figure 1, the b and M values computed may also be used for the subsequent Code-Excited coding of the error signal derived from subtracting a reconstructed residual from a long term residual.

The process may be considered as including the following steps :

30 The $s(n)$ signal is converted into a short-term residual $r(n)$ through a short-term filtering operation using a digital filter with $a(i)$ coefficients. Said coefficients are signal dependent coefficients derived from a pre-emphasized signal $sp(n)$ through short-term analysis operations.

The short-term residual signal $r(n)$ is converted into a first long-term residual $e(n)$, with :

40 wherein : b is a gain factor derived from the short-term residual analysis, M is a pitch multiple; and $r_1(n-M)$ is derived from a reconstructed previous long-term residual, delayed by M.

The first long-term residual signal is coded into a first codeword table address (k_1) and a first gain factor (g_1). This is achieved by correlating a predetermined length block of $e(n)$ samples with pre-stored codewords to determine the address k_1 of the codeword best matching said block.

A coding error signal $r'(n)$ is derived by subtracting a decoded $e_1(n)$ from the uncoded $e(n)$.

The error signal is in turn converted into an error residual $e'(n)$ through a second long-term residual

operation similar to the previous one, i.e. using the already computed M and b coefficients to derive :

$$e'(n) = r'(n) - b \cdot r_2(n-M).$$

- 5 (needless to mention that keeping for this second step the previously computed b and M coefficients helps saving in computing workload. Recomputing these might also be considered).

(6) Second Code-Excited Coding :

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The error residual signal is in turn submitted to Code-Excited coding providing a best matching second codeword address (k2) and a second gain factor (g2).

- The above process provides the data a_i , b's, M's, (g1, k1)'s and (g2, k2)'s to be inserted into a bi-rate frame using conventional multiplexing approaches. Obviously, the process may be extended further to a higher number of rates by repeating the three last steps to generate (g3, k3)'s, (g4, k4)'s, etc, ...

- 15 Synthesizing back the original voice signal from the multi-rate (bi-rate) frame may be achieved as shown in the algorithm of figure 6, assuming the various data had previously been separated from each other through a conventional demultiplexing operation. The k1 and k2 values are used to address a table, set as mentioned above in connection with the coder's description, to fetch the codewords CB(k1) and CB(k2) therefrom. These operations enable reconstructing :

$$e1(n) = g1 \cdot CB(k1, n)$$

$$e2(n) = g2 \cdot CB(k2, n)$$

- 25 Then $e''(n) = e1(n) + e2(n)$

Said $e''(n)$ is then fed into a long-term synthesis filter $1/B(z)$ tuned with b and M and providing $r''(n)$.

$r''(n)$ is then filtered by a short-term synthesis digital filter $1/A(z)$ tuned with the set of a_i coefficients, and providing the synthesized voice signal $s''(n)$.

- A block diagram arrangement of the above synthesizer (receiver) is represented in figure 7. A demultiplexor (60), separates the data from each other. k1 and k2 are used to address the tables (61) and (62), the output of which are fed into multipliers (63) and (64) providing $e1(n)$ and $e2(n)$. An adder (65) adds $e1(n)$ to $e2(n)$ and feeds the result into the filter $1/B(z)$ made of adder (67), a variable delay line (68) adjusted to length M, and a multiplier (69). The output of adder (67) is then filtered through a digital filter (70) with coefficients set to a_i and providing the synthesized back voice signal $s''(n)$.

- 35 The multi-rate approach of this invention may be implemented with more sophisticated coding schemes. For instance, it applies to conventional Base-band coders as represented in figure 8. Once the original voice signal $s(n)$ has been processed to derive the short-term residual $r(n)$, it is split into a low frequency bandwidth (LF) signal $rl(n)$ and a high bandwidth (HF) signal $rh(n)$ using a low-pass filter LPF (70) and adder (71). The high bandwidth energy is computed into a device HFE (72) and coded in (73) into a data designated by E. The output of 73 has been labelled (3). Each one of the bandwidths LF and HF signals, i.e. $rl(n)$ and $rh(n)$ is fed into a multirate CE/LTP coder (75), (76) as represented by (A) and (B) blocks of figure 1. Also either separate (b,M) computing devices or a same one will be used for both bandwidths.

Finally, fed into a multiplexer (77) are the following sets of data :

- 45 - PARCOR related coefficients : a_i
 - Pitch or long-term related data : b's and M's
 - High frequency energy data : E's
 - Low bandwidth multi-rate CE/LTP :
 g_1^1 's; k_1^1 's; g_2^1 's; k_2^1 's
 50 - High bandwidth multi-rate CE/LTP :
 g_1^2 's; k_1^2 's; g_2^2 's; k_2^2 's

This approach enables coding at several rates, with sets of data common to all rates, i.e. the a_i , b and M parameters and the remaining data being inserted or not in the output frame according to the following approaches for instance :

- 55 - Full band coder with a bit rate of 16 Kbps : add
 g_1^1 ; k_1^1 ; g_2^1 ; k_2^1 ; g_1^2 ; k_1^2 ; g_2^2 ; and k_2^2 .
 - Medium band coder :
 g_1^1 ; k_1^1 ; g_2^1 ; k_2^1 ; g_1^2 and k_1^2 only.

- Low band coder :
 $g_1 : k_1 : g_2 : k_2$; and E
- Lower rate coder :
 $g_1 : k_1 : E$.

Obviously, other types of combinations of outputs (1), (2) and (3), a_i , b, M and E might be considered without departing from the scope of this invention.

Claims

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1. A process for multirate coding a voice originating signal using Code-Excited techniques wherein the voice originating signal is considered by blocks of samples and each block is subsequently converted into a prestored table address k and a gain factor g, said multirate process including :

-first Code-Excited coding said voice originating block into a first table address k_1 and a gain g_1 ;

15 - decoding said first Code-Excited coded block;

- subtracting said decoded block from a non-coded voice originating block to derive an error signal block therefrom;

- second Code-Excited coding said error signal block into a second table address k_2 and a gain g_2 ; and

- multiplexing both (g_1, k_1) and (g_2, k_2) data into a single full rate frame;

20 whereby coding at a lower predetermined rate is achieved by simply dropping g_2 and k_2 from the considered frame.

2. A process for multirate encoding a voice originating signal according to claim 1 wherein said voice originating signal is represented by a residual signal derived from the original voice signal to be coded by filtering said original voice signal through a self adjusted short-term filtering operation.

25 3. A process for multirate encoding a voice signal according to claim 2, wherein said short-term filtering is tuned using PARCOR derived coefficients a_i 's computed using a pre-emphasized voice signal.

4. A process according to any one of claims 1-3 wherein said Code-Excited coding involves first subtracting a Long-Term Predicted decoded signal from the residual signal, and then Code-Excited coding the difference.

30 5. A device for multi-rate digitally coding a voice signal $s(n)$ including:

- computing means (10,12) for pre-emphasizing $s(n)$ and deriving from said pre-emphasized $s(n)$, autocorrelation derived coefficients a_i ;

- short-term filtering means (13) tuned by said a_i coefficients and connected to filter $s(n)$ into a short-term residual $r(n)$;

35 - a first Code-Excited coding means including :

- first subtracting means having a (+) input fed with said residual $r(n)$ and providing a long-term residual $e(n)$;

- Code-Excited coding means (15) for converting blocks of $e(n)$ samples into a first table address k_1 and a first gain g_1 ;

40 - decoding means (16) connected to said Code-Excited coding means;

- inverse Long-Term Predictive filtering means (14) connected to said decoding means, the output of said filtering means (14) being fed to the (-) input of said first subtracting means;

- long-term computing means filter (11) connected to said short-term filtering means and to said inverse Long-Term Predictive means to provide b and M factors for tuning said filter (14);

45 - second subtracting means (17) having a (+) input connected to receive said long-term residual $e(n)$ and a (-) input connected to said decoding means (16), said subtracting means (17) providing an error signal $r'(n)$;

- second Code-Excited coding means similar to said first Code-Excited coding means, fed with said error signal $r'(n)$ and providing second table address k_2 and gain g_2 ;

- multiplexing means for multiplexing a_i 's; b's; M's; (g_1, k_1) 's and (g_2, k_2) 's into a single full rate frame.

50 6. A device for decoding the signal digitally coded by the coder according to claim 5, said decoder including :

- demultiplexing means for separating a_i , b's, M's, g_1 's, k_1 's, g_2 's and k_2 's from each other;

- table means (61-62) addressed with k_1 and k_2 ;

- multiplier means (63-64) connected to said table means and multiplying said tables outputs by g_1 , and g_2 respectively;

55 - first adding means (65) connected to said multipliers output.

- second adding means (67) having a first input connected to first adding means, and a second input fed with said second adding means output through a delay line adjusted to M and a multiplier by b; and,
- short-term inverse filtering means (70) tuned with a_i 's coefficients and connected to said second adder.

7. A base-band multi-rate coder for coding a voice signal according to claim 5 wherein said residual
 5 signal is split into a low frequency bandwidth signal $rl(n)$ and a high frequency bandwidth signal $rh(n)$, said
 $rh(n)$ and $rl(n)$ being subsequently multirate encoded into couples (g_1^1, k_1^1) 's, (g_2^1, k_2^1) 's, (g_1^2, k_1^2) 's and (g_2^2, k_2^2) 's.

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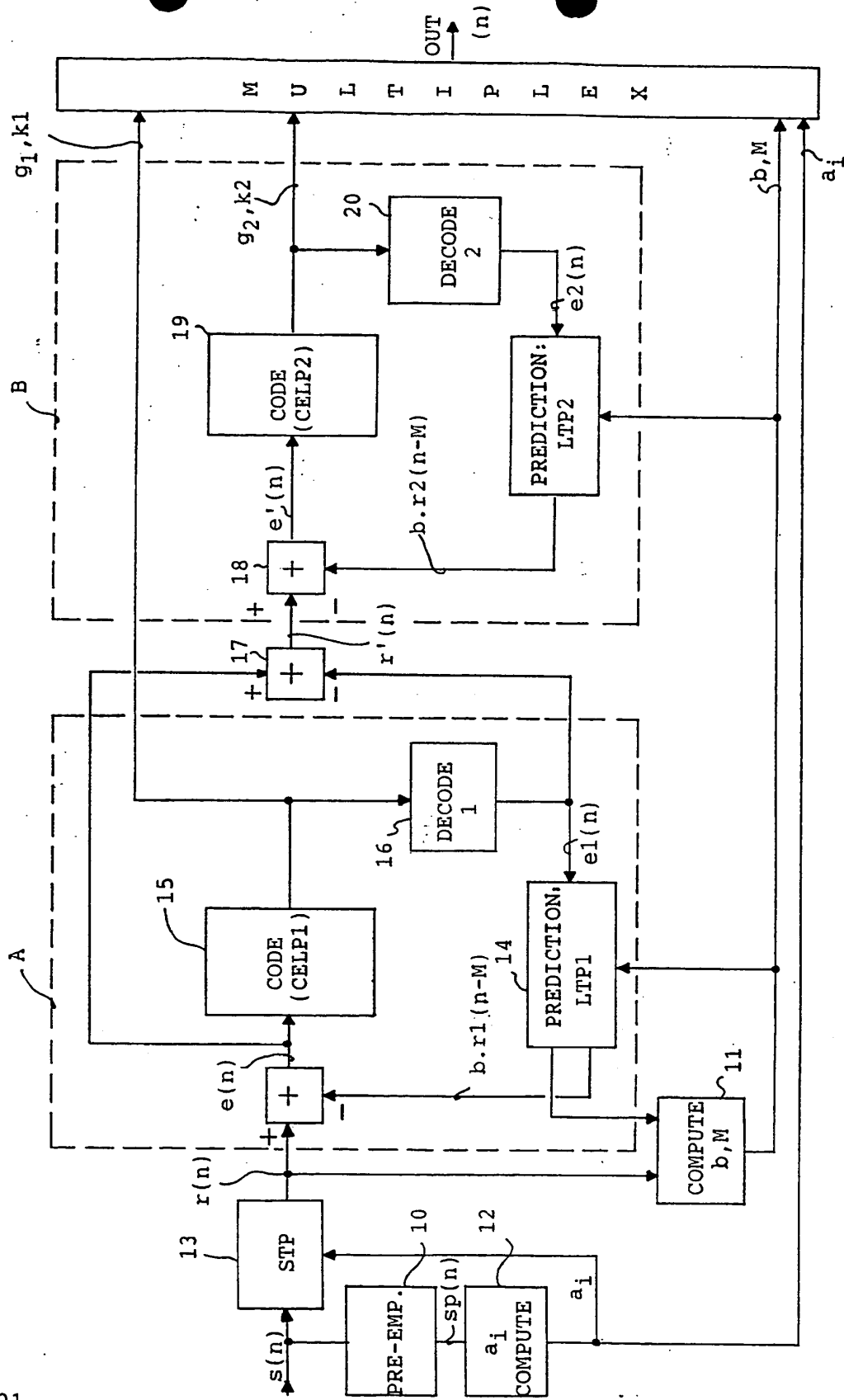
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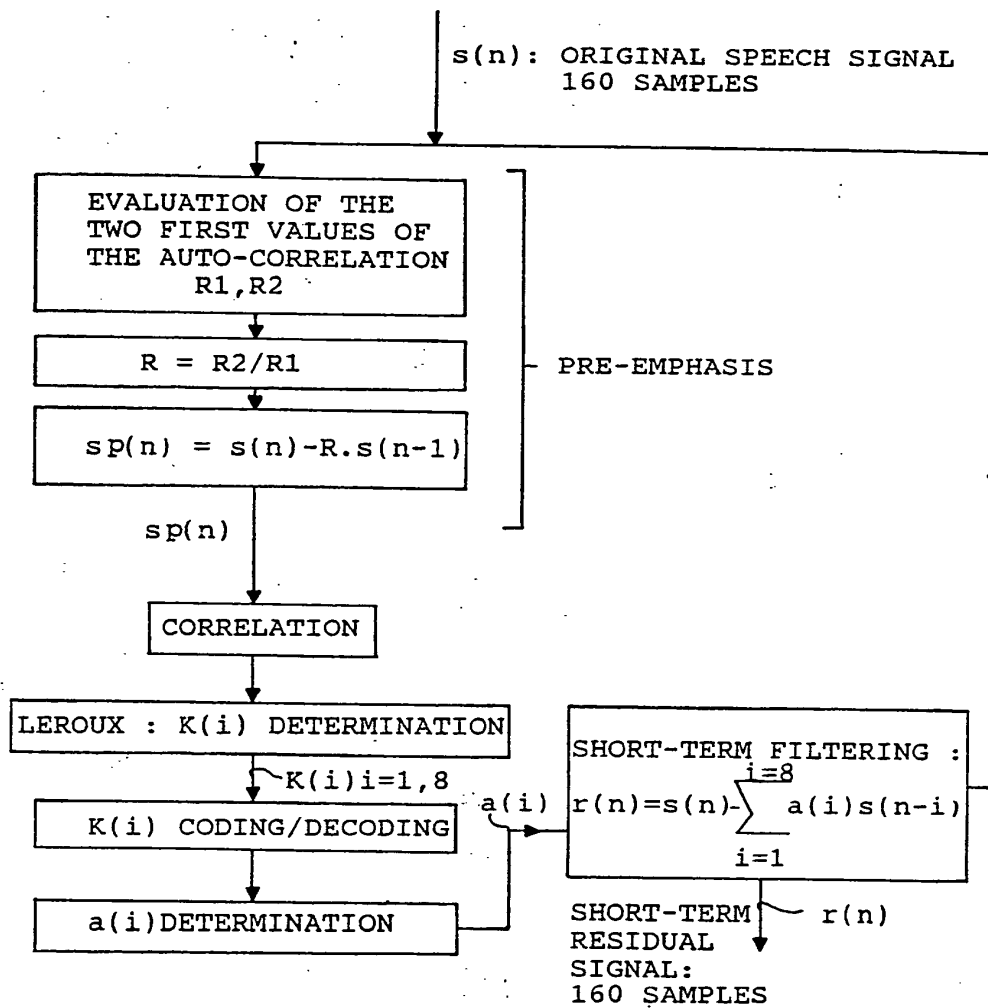
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FIG. 1



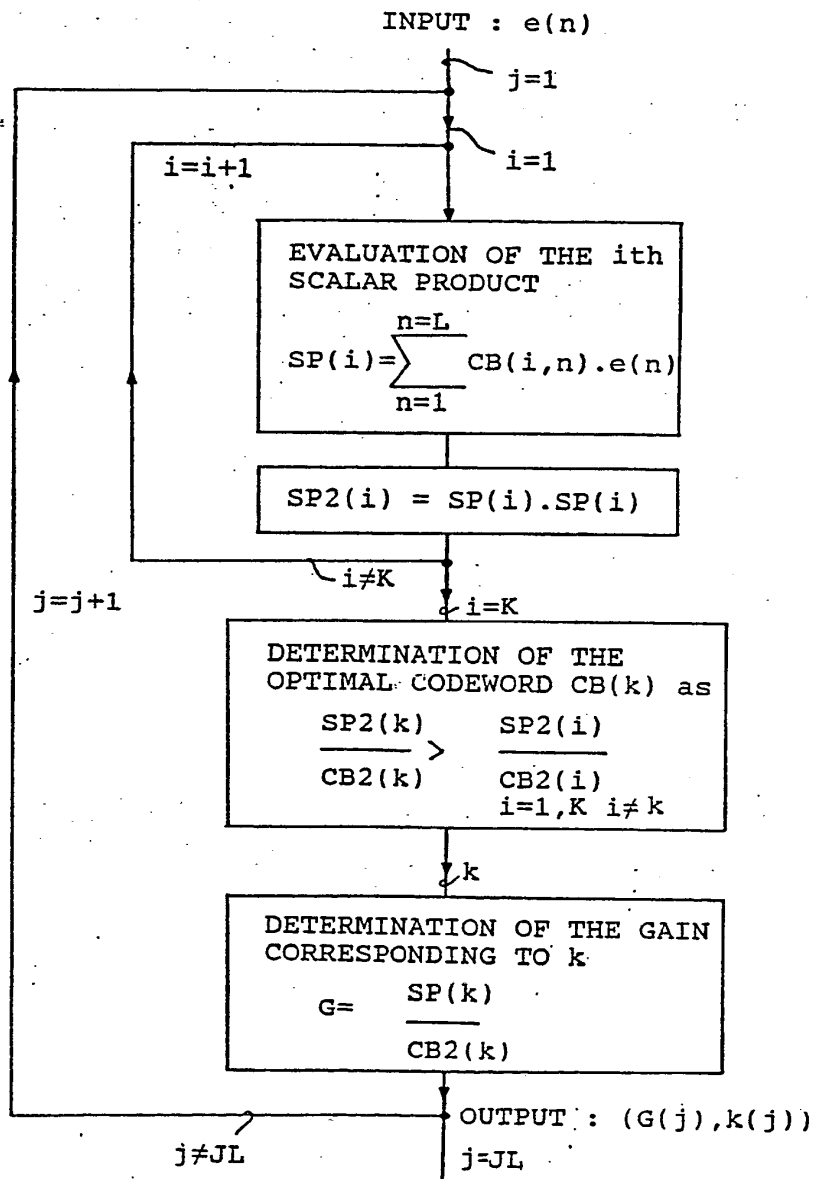
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FIG. 2



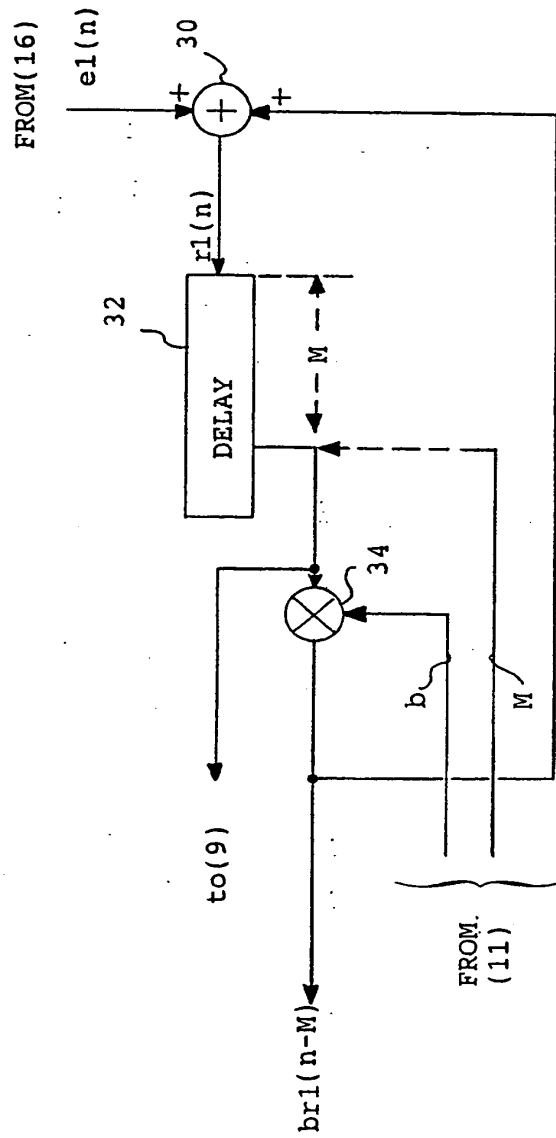
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FIG. 3



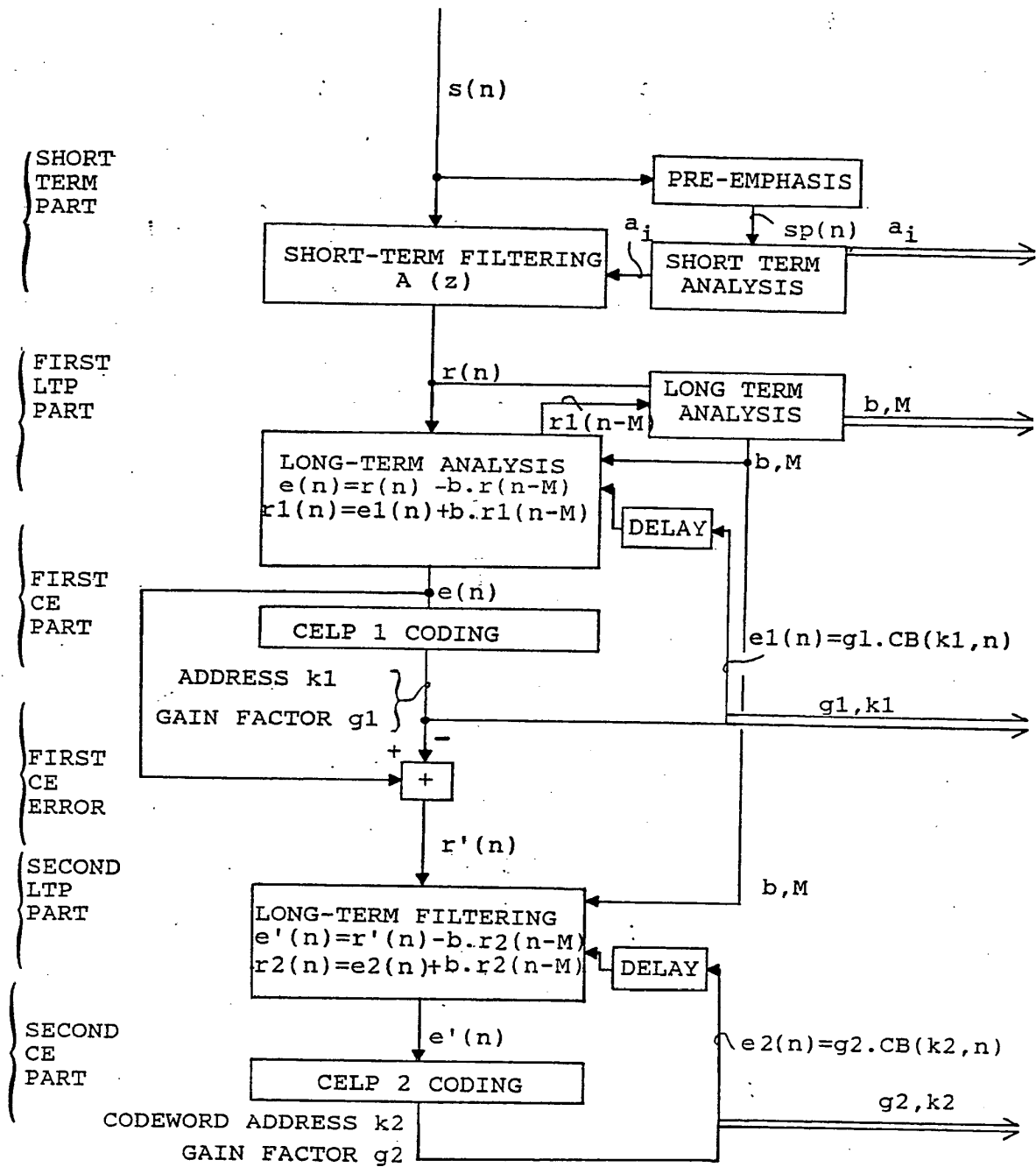
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FIG. 4



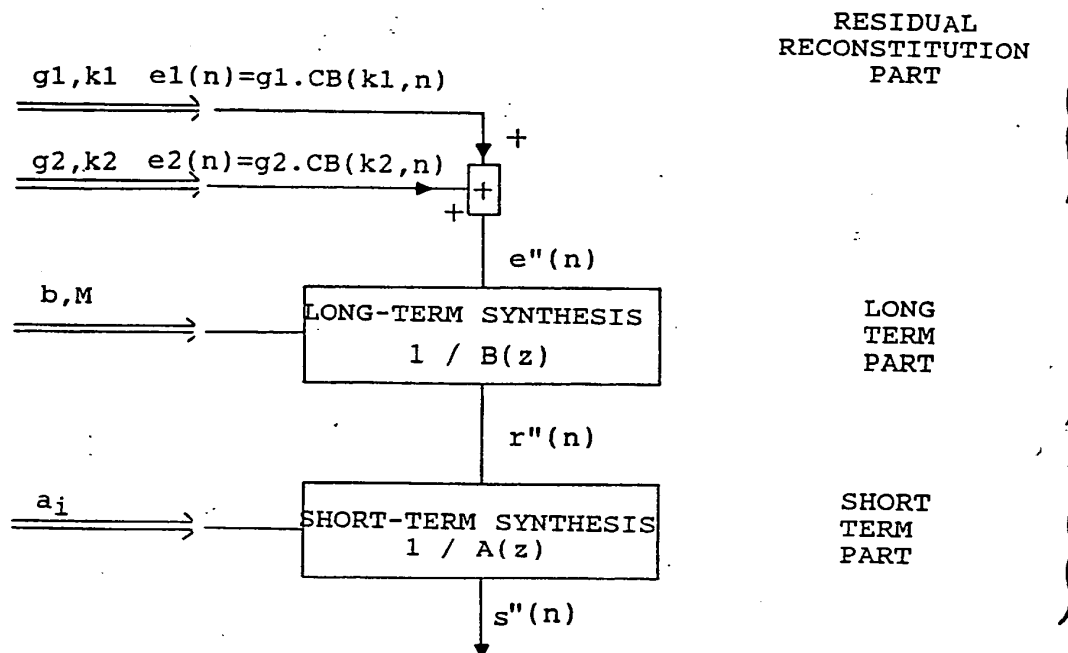
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FIG. 5



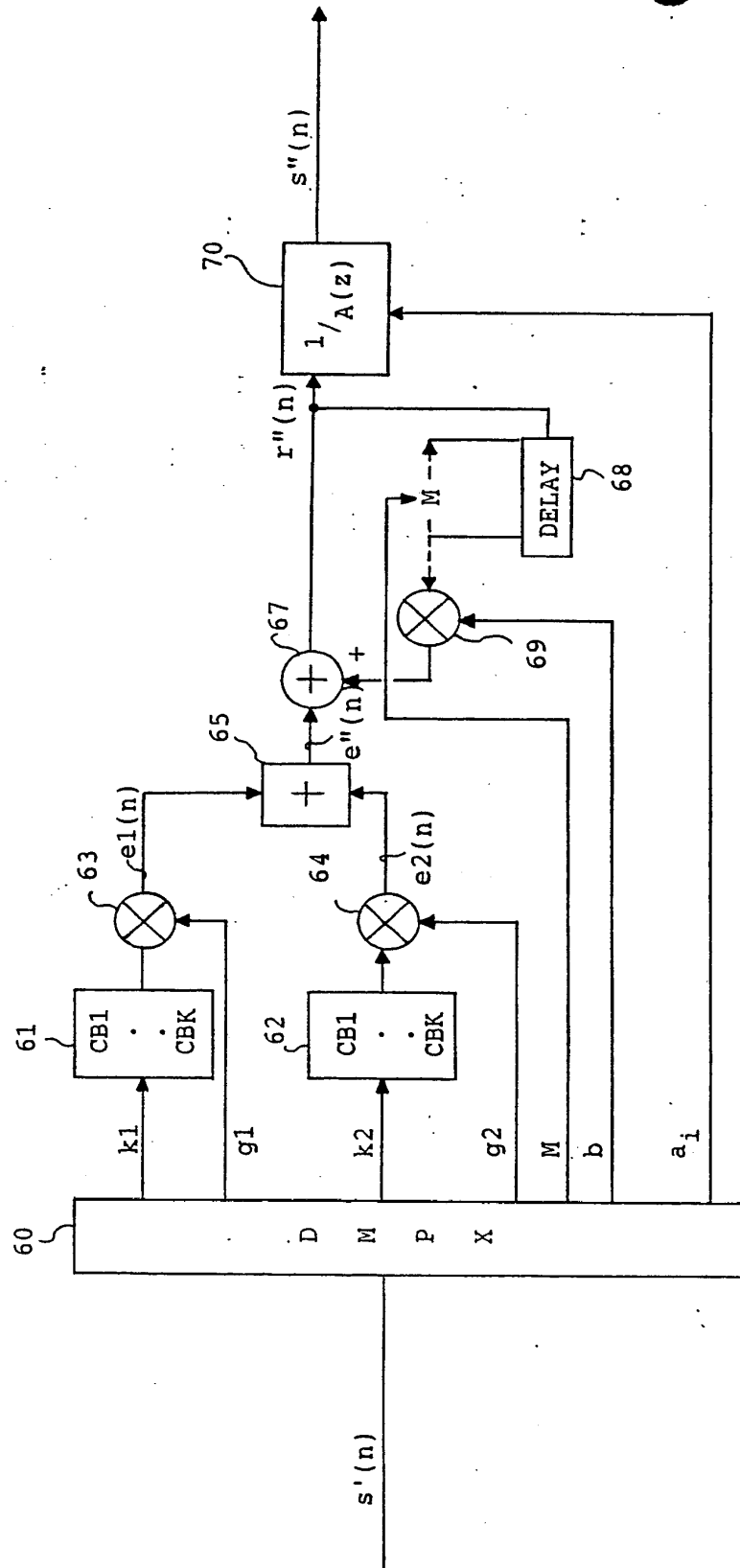
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FIG. 6



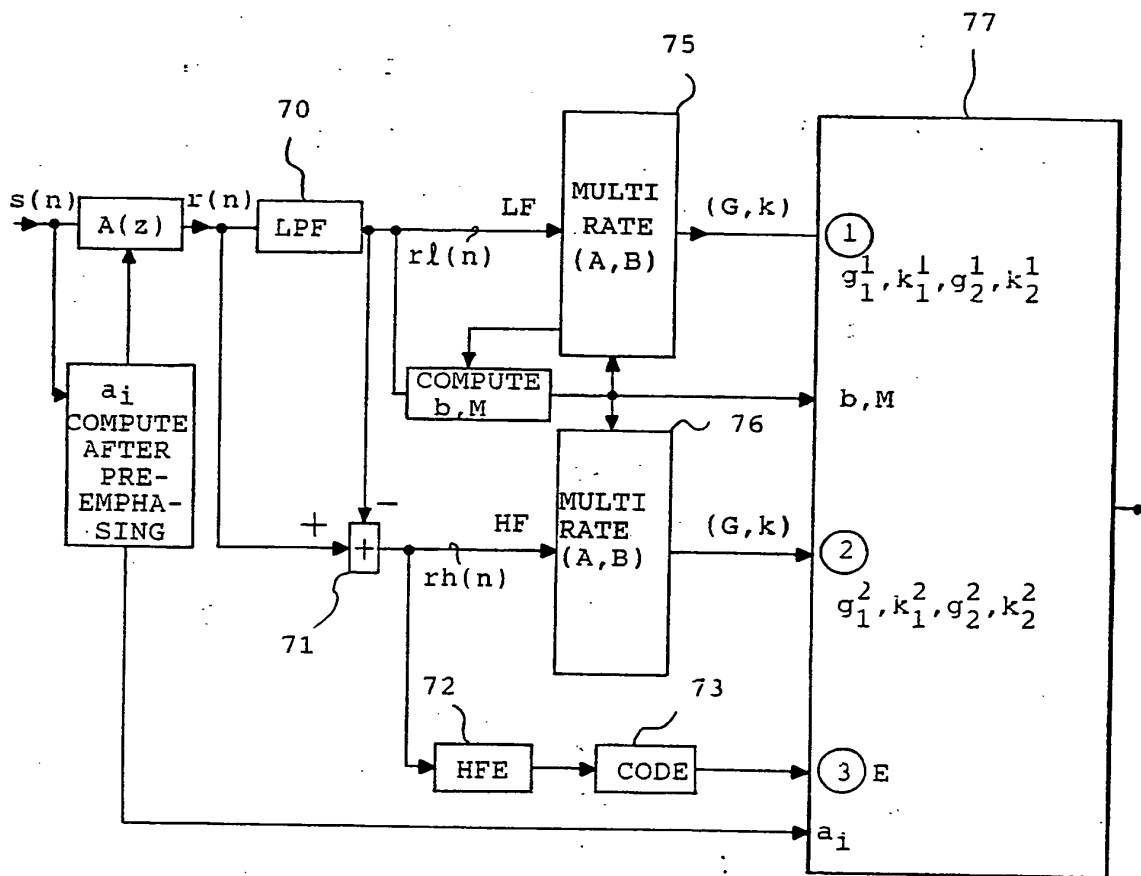
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FIG. 7



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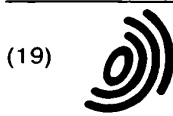
EUROPEAN SEARCH REPORT

Application Number

EP 88 48 0007

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.4)
A	ICASSP 85, PROCEEDINGS OF THE IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH, AND SIGNAL PROCESSING, Tampa, 26th-29th March 1985, vol. 4, pages 1703-1706, IEEE, New York, US; A. HAQUI et al.: "Embedded coding of speech: a vector quantization approach" * Paragraph 4.2: "Multistage coders" * ---	1	G 10 L 9/14
A	IBM TECHNICAL DISCLOSURE BULLETIN, vol. 29, no. 2, July 1986, pages 929,930, New York, US; "Multipulse excited linear predictive coder" * Figure 1 * ---	2,4,5	
A	ICASSP 83, PROCEEDINGS OF THE IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH AND SIGNAL PROCESSING, Boston, 14th-16th April 1983, vol. 3, pages 1312-1315, IEEE, New York, US; L. BERTORELLO et al.: "Design of a 4.8/9.6 Kbps baseband LPC coder using split-band and vector quantization" * Paragraph 2.2: "VQ of the baseband" * -----	7	
			TECHNICAL FIELDS SEARCHED (Int. Cl.4)
			G 10 L 9/14
The present search report has been drawn up for all claims			
Place of search THE HAGUE		Date of completion of the search 23-11-1988	Examiner ARMSPACH J.F.A.M.
CATEGORY OF CITED DOCUMENTS			
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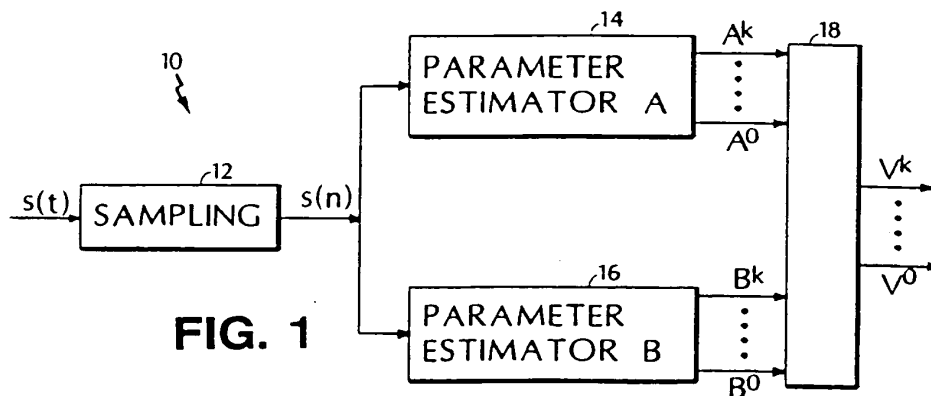
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(54) Estimation of excitation parameters

(57) Excitation parameters for a digitized speech signal are determined by analysing the digitized speech signal. The digitized speech signal is divided into at least two frequency bands. A first preliminary excitation parameter is determined by performing a nonlinear operation on at least one of the frequency band signals to produce a modified frequency band signal and determining the first preliminary excitation parameter using the modified frequency band signal. A second preliminary

excitation parameter is determined using a method different from the first method. The first and second preliminary excitation parameters are used to determine an excitation parameter for the digitized speech signal. The method is useful in encoding speech. Speech synthesized using the parameters estimated based on the invention generates high quality speech at various bit rates useful for applications such as satellite voice communication.



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CORRIGENDUM

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Description

The invention has arisen from work seeking to improve the accuracy with which excitation parameters are estimated in speech analysis and synthesis.

Speech analysis and synthesis are widely used in applications such as telecommunications and voice recognition. A vocoder, which is a type of speech analysis/synthesis system, models speech as the response of a system to excitation over short time intervals. Examples of vocoder systems include linear prediction vocoders, homomorphic vocoders, channel vocoders, sinusoidal transform coders ("STC"), multiband excitation ("MBE") vocoders, improved multiband excitation ("IMBE (TM)") vocoders.

Vocoders typically synthesize speech based on excitation parameters and system parameters. Typically, an input signal is segmented using, for example, a Hamming window. Then, for each segment, system parameters and excitation parameters are determined. System parameters include the spectral envelope or the impulse response of the system. Excitation parameters include a fundamental frequency (or pitch) and a voiced/unvoiced parameter that indicates whether the input signal has pitch (or indicates the degree to which the input signal has pitch). In vocoders that divide the speech into frequency bands, such as IMBE (TM) vocoders, the excitation parameters may also include a voiced/unvoiced parameter for each frequency band rather than a single voiced/unvoiced parameter. Accurate excitation parameters are essential for high quality speech synthesis.

When the voiced/unvoiced parameters include only a single voiced/unvoiced decision for the entire frequency band, the synthesized speech tends to have a "buzzy" quality especially noticeable in regions of speech which contain mixed voicing or in voiced regions of noisy speech. A number of mixed excitation models have been proposed as potential solutions to the problem of "buziness" in vocoders. In these models, periodic and noise-like excitations are mixed which have either time-invariant or time-varying spectral shapes.

In excitation models having time-invariant spectral shapes, the excitation signal consists of the sum of a periodic source and a noise source with fixed spectral envelopes. The mixture ratio controls the relative amplitudes of the periodic and noise sources. Examples of such models include Itakura and Saito, "Analysis Synthesis Telephony Based upon the Maximum Likelihood Method," Reports of 6th Int. Cong. Acoust., Tokyo, Japan, Paper C-5-5, pp. C17-20, 1968; and Kwon and Goldberg, "An Enhanced LPC Vocoder with No Voiced/Unvoiced Switch," IEEE Trans. on Acoust., Speech, and Signal Processing, vol. ASSP-32, no. 4, pp. 851-858, August 1984. In these excitation models a white noise source is added to a white periodic source. The mixture ratio between these sources is estimated from the height of the peak of the autocorrelation of the LPC residual.

In excitation models having time-varying spectral shapes, the excitation signal consists of the sum of a periodic source and a noise source with time varying spectral envelope shapes. Examples of such models include Fujimara, "An Approximation to Voice Aperiodicity," IEEE Trans. Audio and Electroacoust., pp. 68-72, March 1968; Makhoul et al., "A Mixed-Source Excitation Model for Speech Compression and Synthesis," IEEE Int. Conf. on Acoust. Sp. & Sig. Proc., April 1978, pp. 163-166; Kwon and Goldberg, "An Enhanced LPC Vocoder with No Voiced/Unvoiced Switch," IEEE Trans. on Acoust., Speech, and Signal Processing, vol. ASSP-32, no. 4, pp. 851-858, August 1984; and Griffin and Lim, "Multiband Excitation Vocoder," IEEE Trans. Acoust., Speech, Signal Processing, vol. ASSP-36, pp. 1223-1235, Aug. 1988.

In the excitation model proposed by Fujimara, the excitation spectrum is divided into three fixed frequency bands. A separate cepstral analysis is performed for each frequency band and a voiced/unvoiced decision for each frequency band is made based on the height of the cepstrum peak as a measure of periodicity.

In the excitation model proposed by Makhoul et al., the excitation signal consists of the sum of a low-pass periodic source and a high-pass noise source. The low-pass periodic source is generated by filtering a white pulse source with a variable cut-off low-pass filter. Similarly, the high-pass noise source was generated by filtering a white noise source with a variable cut-off high-pass filter. The cut-off frequencies for the two filters are equal and are estimated by choosing the highest frequency at which the spectrum is periodic. Periodicity of the spectrum is determined by examining the separation between consecutive peaks and determining whether the separations are the same, within some tolerance level.

In a second excitation model implemented by Kwon and Goldberg, a pulse source is passed through a variable gain low-pass filter and added to itself, and a white noise source is passed through a variable gain high-pass filter and added to itself. The excitation signal is the sum of the resultant pulse and noise sources with the relative amplitudes controlled by a voiced/unvoiced mixture ratio. The filter gains and voiced/unvoiced mixture ratio are estimated from the LPC residual signal with the constraint that the spectral envelope of the resultant excitation signal is flat.

In the multiband excitation model proposed by Griffin and Lim, a frequency dependent voiced/unvoiced mixture function is proposed. This model is restricted to a frequency dependent binary voiced/unvoiced decision for coding purposes. A further restriction of this model divides the spectrum into a finite number of frequency bands with a binary voiced/unvoiced decision for each band. The voiced/unvoiced information is estimated by comparing the speech spectrum to the closest periodic spectrum. When the error is below a threshold, the band is marked voiced, otherwise, the

band is marked unvoiced.

Excitation parameters may also be used in applications, such as speech recognition, where no speech synthesis is required. Once again, the accuracy of the excitation parameters directly affects the performance of such a system.

In one aspect, generally, the invention features a hybrid excitation parameter estimation technique that produces two sets of excitation parameters for a speech signal using two different approaches and combines the two sets to produce a single set of excitation parameters. In a first approach, the technique applies a nonlinear operation to the speech signal to emphasize the fundamental frequency of the speech signal. In a second approach, we use a different method which may or may not include a nonlinear operation. While the first approach produces highly accurate excitation parameters under most conditions, the second approach produces more accurate parameters under certain conditions. By using both approaches and combining the resulting sets of excitation parameters to produce a single set, our technique produces accurate results under a wider range of conditions than are produced by either of the approaches individually.

In typical approaches to determining excitation parameters, an analog speech signal $s(t)$ is sampled to produce a speech signal $s(n)$. Speech signal $s(n)$ is then multiplied by a window $w(n)$ to produce a windowed signal $s_w(n)$ that is commonly referred to as a speech segment or a speech frame. A Fourier transform is then performed on windowed signal $s_w(n)$ to produce a frequency spectrum $S_w(\omega)$ from which the excitation parameters are determined.

When speech signal $s(n)$ is periodic with a fundamental frequency ω_0 or pitch period n_0 (where n_0 equals $2\pi/\omega_0$), the frequency spectrum of speech signal $s(n)$ should be a line spectrum with energy at ω_0 and harmonics thereof (integral multiples of ω_0). As expected, $S_w(\omega)$ has spectral peaks that are centered around ω_0 and its harmonics. However, due to the windowing operation, the spectral peaks include some width, where the width depends on the length and shape of window $w(n)$ and tends to decrease as the length of window $w(n)$ increases. This window-induced error reduces the accuracy of the excitation parameters. Thus, to decrease the width of the spectral peaks, and to thereby increase the accuracy of the excitation parameters, the length of window $w(n)$ should be made as long as possible.

The maximum useful length of window $w(n)$ is limited. Speech signals are not stationary signals, and instead have fundamental frequencies that change over time. To obtain meaningful excitation parameters, an analyzed speech segment must have a substantially unchanged fundamental frequency. Thus, the length of window $w(n)$ must be short enough to ensure that the fundamental frequency will not change significantly within the window.

In addition to limiting the maximum length of window $w(n)$, a changing fundamental frequency tends to broaden the spectral peaks. This broadening effect increases with increasing frequency. For example, if the fundamental frequency changes by $\Delta\omega_0$ during the window, the frequency of the m th harmonic, which has a frequency of $m\omega_0$, changes by $m\Delta\omega_0$ so that the spectral peak corresponding to $m\omega_0$ is broadened more than the spectral peak corresponding to ω_0 . This increased broadening of the higher harmonics reduces the effectiveness of higher harmonics in the estimation of the fundamental frequency and the generation of voiced/unvoiced parameters for high frequency bands.

By applying a nonlinear operation to the speech signal, the increased impact on higher harmonics of a changing fundamental frequency is reduced or eliminated, and higher harmonics perform better in estimation of the fundamental frequency and determination of voiced/unvoiced parameters. Suitable nonlinear operations map from complex (or real) to real values and produce outputs that are nondecreasing functions of the magnitudes of the complex (or real) values. Such operations include, for example, the absolute value, the absolute value squared, the absolute value raised to some other power, or the log of the absolute value.

Nonlinear operations tend to produce output signals having spectral peaks at the fundamental frequencies of their input signals. This is true even when an input signal does not have a spectral peak at the fundamental frequency. For example, if a bandpass filter that only passes frequencies in the range between the third and fifth harmonics of ω_0 is applied to a speech signal $s(n)$, the output of the bandpass filter, $x(n)$, will have spectral peaks at $3\omega_0$, $4\omega_0$ and $5\omega_0$.

Though $x(n)$ does not have a spectral peak at ω_0 , $|x(n)|^2$ will have such a peak. For a real signal $x(n)$, $|x(n)|^2$ is equivalent to $x^2(n)$. As is well known, the Fourier transform of $x^2(n)$ is the convolution of $X(\omega)$, the Fourier transform of $x(n)$, with $X(\omega)$:

$$\sum_{n=-\infty}^{\infty} x^2(n) e^{-j\omega n} = \frac{1}{2\pi} \int_{-\pi}^{\pi} X(\omega-u) X(u) du.$$

The convolution of $X(\omega)$ with $X(\omega)$ has spectral peaks at frequencies equal to the differences between the frequencies for which $X(\omega)$ has spectral peaks. The differences between the spectral peaks of a periodic signal are the fundamental frequency and its multiples. Thus, in the example in which $X(\omega)$ has spectral peaks at $3\omega_0$, $4\omega_0$ and $5\omega_0$, $X(\omega)$ convolved with $X(\omega)$ has a spectral peak at ω_0 ($4\omega_0-3\omega_0$, $5\omega_0-4\omega_0$). For a typical periodic signal, the spectral peak at the funda-

mental frequency is likely to be the most prominent.

The above discussion also applies to complex signals. For a complex signal $x(n)$, the Fourier transform of $|x(n)|^2$ is:

$$\sum_{n=-\infty}^{\infty} |x(n)|^2 e^{-j\omega n} = \frac{1}{2\pi} \int_{-\pi}^{\pi} X(\omega+u) X^*(u) du.$$

This is an autocorrelation of $X(\omega)$ with $X^*(\omega)$, and also has the property that spectral peaks separated by $n\omega_0$ produce peaks at $n\omega_0$.

Even though $|x(n)|$, $|x(n)|^a$ for some real "a", and $\log |x(n)|$ are not the same as $|x(n)|^2$, the discussion above for $|x(n)|^2$ applies approximately at the qualitative level. For example, for $|x(n)| = y(n)^{0.5}$, where $y(n) = |x(n)|^2$, a Taylor series expansion of $y(n)$ can be expressed as:

$$|x(n)| = \sum_{k=0}^{\infty} c_k y^k(n).$$

Because multiplication is associative, the Fourier transform of the signal $y^k(n)$ is $Y(\omega)$ convolved with the Fourier transform of $y^{k-1}(n)$. The behavior for nonlinear operations other than $|x(n)|^2$ can be derived from $|x(n)|^2$ by observing the behavior of multiple convolutions of $Y(\omega)$ with itself. If $Y(\omega)$ has peaks at $n\omega_0$, then multiple convolutions of $Y(\omega)$ with itself will also have peaks at $n\omega_0$.

As shown, nonlinear operations emphasize the fundamental frequency of a periodic signal, and are particularly useful when the periodic signal includes significant energy at higher harmonics. However, the presence of the nonlinearity can degrade performance in some cases. For example, performance may be degraded when speech signal $s(n)$ is divided into multiple bands $s^i(n)$ using bandpass filters, where $s^i(n)$ denotes the result of bandpass filtering using the i th bandpass filter. If a single harmonic of the fundamental frequency is present in the pass band of the i th filter, the output of the filter is:

$$s^i(n) = A_k e^{j(\omega_k n + \theta_k)}$$

where ω_k is the frequency, θ_k is the phase, and A_k is the amplitude of the harmonic. When a nonlinearity such as the absolute value is applied to $s^i(n)$ to produce a value $y^i(n)$, the result is:

$$y^i(n) = |s^i(n)| = |A_k|$$

so that the frequency information has been completely removed from the signal $y^i(n)$. Removal of this frequency information can reduce the accuracy of parameter estimates.

Our hybrid technique provides significantly improved parameter estimation performance in cases for which the nonlinearity reduces the accuracy of parameter estimates while maintaining the benefits of the nonlinearity in the remaining cases. As described above, the hybrid technique includes combining parameter estimates based on the signal after the nonlinearity has been applied ($y^i(n)$) with parameter estimates based on the signal before the nonlinearity is applied ($s^i(n)$ or $s(n)$). The two approaches produce parameter estimates along with an indication of the probability of correctness of these parameter estimates. The parameter estimates are then combined giving higher weight to estimates with a higher probability of being correct.

In another aspect, generally, the invention features the application of smoothing techniques to the voiced/unvoiced parameters. Voiced/unvoiced parameters can be binary or continuous functions of time and/or frequency. Because these parameters tend to be smooth functions in at least one direction (positive or negative) of time or frequency, the estimates of these parameters can benefit from appropriate application of smoothing techniques in time and/or frequency.

The invention also features an improved technique for estimating voiced/unvoiced parameters. In vocoders such as linear prediction vocoders, homomorphic vocoders, channel vocoders, sinusoidal transform coders, multiband excitation vocoders, and IMBE (TM) vocoders, a pitch period n (or equivalently a fundamental frequency) is selected.

Thereafter, a function $f'(n)$ is then evaluated at the selected pitch period (or fundamental frequency) to estimate the i th voiced/unvoiced parameter. However, for some speech signals, evaluation of this function only at the selected pitch period will result in reduced accuracy of one or more voiced/unvoiced parameter estimates. This reduced accuracy may result from speech signals that are more periodic at a multiple of the pitch period than at the pitch period, and may be frequency dependent so that only certain portions of the spectrum are more periodic at a multiple of the pitch period. Consequently, the voiced/unvoiced parameter estimation accuracy can be improved by evaluating the function $f'(n)$ at the pitch period n and at its multiples, and thereafter combining the results of these evaluations.

In another aspect, the invention features an improved technique for estimating the fundamental frequency or pitch period. When the fundamental frequency ω_0 (or pitch period n_0) is estimated, there may be some ambiguity as to whether ω_0 or a submultiple or multiple of ω_0 is the best choice for the fundamental frequency. Since the fundamental frequency tends to be a smooth function of time for voiced speech, predictions of the fundamental frequency based on past estimates can be used to resolve ambiguities and improve the fundamental frequency estimate.

Other features and advantages of the invention will be apparent from the following description of the preferred embodiments, given by way of example only.

In the drawings:

Fig. 1 is a block diagram of a system for determining whether frequency bands of a signal are voiced or unvoiced.

Fig. 2 is a block diagram of a parameter estimation unit of the system of Fig. 1.

Fig. 3 is a block diagram of a channel processing unit of the parameter estimation unit of Fig. 2.

Fig. 4 is a block diagram of a parameter estimation unit of the system of Fig. 1.

Fig. 5 is a block diagram of a channel processing unit of the parameter estimation unit of Fig. 4.

Fig. 6 is a block diagram of a parameter estimation unit of the system of Fig. 1.

Fig. 7 is a block diagram of a channel processing unit of the parameter estimation unit of Fig. 6.

Figs. 8-10 are block diagrams of systems for determining the fundamental frequency of a signal.

Fig. 11 is a block diagram of voiced/unvoiced parameter smoothing unit.

Fig. 12 is a block diagram of voiced/unvoiced parameter improvement unit.

Fig. 13 is a block diagram of a fundamental frequency improvement unit.

Figs. 1-12 show the structure of a system for estimating excitation parameters, the various blocks and units of which are preferably implemented with software.

With reference to Fig. 1, a voiced/unvoiced determination system 10 includes a sampling unit 12 that samples an analog speech signal $s(t)$ to produce a speech signal $s(n)$. For typical speech coding applications, the sampling rate ranges between six kilohertz and ten kilohertz.

Speech signal $s(n)$ is supplied to a first parameter estimator 14 that divides the speech signal into $k+1$ bands and produces a first set of preliminary voiced/unvoiced ("V/UV") parameters (A^0 to A^K) corresponding to a first estimate as to whether the signals in the bands are voiced or unvoiced. Speech signal $s(n)$ is also supplied to a second parameter estimator 16 that produces a second set of preliminary V/UV parameters (B^0 to B^K) that correspond to a second estimate as to whether the signals in the bands are voiced or unvoiced. The two sets of preliminary V/UV parameters are combined by a combination block 18 to produce a set of V/UV parameters (V^0 to V^K).

With reference to Fig. 2, first parameter estimator 14 produces the first voiced/unvoiced estimate using a frequency domain approach. Channel processing units 20 in first parameter estimator 14 divide speech signal $s(n)$ into at least two frequency bands and process the frequency bands to produce a first set of frequency band signals, designated as $T^0(\omega) \dots T^1(\omega)$. As discussed below, channel processing units 20 are differentiated by the parameters of a bandpass filter used in the first stage of each channel processing unit 20. In the described embodiment, there are sixteen channel processing units (I equals 15).

A remap unit 22 transforms the first set of frequency band signals to produce a second set of frequency band signals, designated as $U^0(\omega) \dots U^K(\omega)$. In the described embodiment, there are eight frequency band signals in the second set of frequency band signals (K equals 7). Thus, remap unit 22 maps the frequency band signals from the sixteen channel processing units 20 into eight frequency band signals. Remap unit 20 does so by combining consecutive pairs of frequency band signals from the first set into single frequency band signals in the second set. For example, $T^0(\omega)$ and $T^1(\omega)$ are combined to produce $U^0(\omega)$, and $T^{14}(\omega)$ and $T^{15}(\omega)$ are combined to produce $U^7(\omega)$. Other approaches to remapping could also be used.

Next, voiced/unvoiced parameter estimation units 24, each associated with a frequency band signal from the second set, produce preliminary V/UV parameters A^0 to A^K by computing a ratio of the voiced energy in the frequency band at an estimated fundamental frequency ω^0 to the total energy in the frequency band and subtracting this ratio from 1:

$$A^k = 1.0 - E_v^k(\omega_0) / E_t^k$$

The voiced energy in the frequency band is computed as:

$$E_V^k(\omega_o) = \sum_{n=1}^N \sum_{\omega_m \in I_n} U^k(\omega_m)$$

where

$$I_n = [(n-0.25) \omega_o, (n+0.25) \omega_o],$$

and N is the number of harmonics of the fundamental frequency ω_o being considered. V/UV parameter estimation units 24 determine the total energy of their associated frequency band signals as:

$$E_T^k = \sum_{\forall \omega_m > 0.5 \omega_o} U^k(\omega_m).$$

The degree to which the frequency band signal is voiced varies indirectly with the value of the preliminary V/UV parameter. Thus, the frequency band signal is highly voiced when the preliminary V/UV parameter is near zero and is highly unvoiced when the parameter is greater than or equal to one half.

With reference to Fig. 3, when speech signal $s(n)$ enters a channel processing unit 20, components $s'(n)$ belonging to a particular frequency band are isolated by a bandpass filter 26. Bandpass filter 26 uses downsampling to reduce computational requirements, and does so without any significant impact on system performance. Bandpass filter 26 can be implemented as a Finite Impulse Response (FIR) or Infinite Impulse Response (IIR) filter, or by using an FFT. In the described embodiment, bandpass filter 26 is implemented using a thirty two point real input FFT to compute the outputs of a thirty two point FIR filter at seventeen frequencies, and achieves a downsampling factor of S by shifting the input by S samples each time the FFT is computed. For example, if a first FFT used samples one through thirty two, a downsampling factor of ten would be achieved by using samples eleven through forty two in a second FFT.

A first nonlinear operation unit 28 then performs a nonlinear operation on the isolated frequency band $s'(n)$ to emphasize the fundamental frequency of the isolated frequency band $s'(n)$. For complex values of $s'(n)$ (i greater than zero), the absolute value, $|s'(n)|$, is used. For the real value of $s^0(n)$, $s^0(n)$ is used if $s^0(n)$ is greater than zero and zero is used if $s^0(n)$ is less than or equal to zero.

The output of nonlinear operation unit 28 is passed through a lowpass filtering and downsampling unit 30 to reduce the data rate and consequently reduce the computational requirements of later components of the system. Lowpass filtering and downsampling unit 30 uses an FIR filter computed every other sample for a downsampling factor of two.

A windowing and FFT unit 32 multiplies the output of lowpass filtering and downsampling unit 30 by a window and computes a real input FFT, $S^i(\omega)$, of the product. Typically, windowing and FFT unit 32 uses a Hamming window and a real input FFT.

Finally, a second nonlinear operation unit 34 performs a nonlinear operation on $S^i(\omega)$ to facilitate estimation of voiced or total energy and to ensure that the outputs of channel processing units 20, $T^i(\omega)$, combine constructively if used in fundamental frequency estimation. The absolute value squared is used because it makes all components of $T^i(\omega)$ real and positive.

With reference to Fig. 4, second parameter estimator 16 produces the second preliminary V/UV estimates using a sinusoid detector/estimator. Channel processing units 36 in second parameter estimator 16 divide speech signal $s(n)$ into at least two frequency bands and process the frequency bands to produce a first set of signals, designated as $R^0(l) \dots R^l(l)$. Channel processing units 36 are differentiated by the parameters of a bandpass filter used in the first stage of each channel processing unit 36. In the described embodiment, there are sixteen channel processing units (l equals 15). The number of channels (value of l) in Fig. 4 does not have to equal the number of channels (value of l) in Fig. 2.

A remap unit 38 transforms the first set of signals to produce a second set of signals, designated as $S^0(l) \dots S^K(l)$. The remap unit can be an identity system. In the described embodiment, there are eight signals in the second set of signals (K equals 7). Thus, remap unit 38 maps the signals from the sixteen channel processing units 36 into eight signals. Remap unit 38 does so by combining consecutive pairs of signals from the first set into single signals in the

second set. For example, $R^0(l)$ and $R^1(l)$ are combined to produce $S^0(l)$, and $R^{14}(l)$ and $R^{15}(l)$ are combined to produce $S^7(l)$. Other approaches to remapping could also be used.

Next, V/UV parameter estimation units 40, each associated with a signal from the second set, produce preliminary V/UV parameters B^0 to B^K by computing a ratio of the sinusoidal energy in the signal to the total energy in the signal and subtracting this ratio from 1:

$$B^k = 1.0 - S^k(1)/S^k(0).$$

With reference to Fig. 5, when speech signal $s(n)$ enters a channel processing unit 36, components $s^i(n)$ belonging to a particular frequency band are isolated by a bandpass filter 26 that operates identically to the bandpass filters of channel processing units 20 (see Fig. 3). It should be noted that, to reduce computation requirements, the same bandpass filters may be used in channel processing units 20 and 36, with the outputs of each filter being supplied to a first nonlinear operation unit 28 of a channel processing unit 20 and a window and correlate unit 42 of a channel processing unit 36.

A window and correlate unit 42 then produces two correlation values for the isolated frequency band $s^i(n)$. The first value, $R^i(0)$, provides a measure of the total energy in the frequency band:

$$R^i(0) = \left[\frac{1}{2} \sum_{n=0}^{N-1} [|s^i(n)|^2 + |s^i(n+S)|^2] \right]^2$$

where N is related to the size of the window and typically defines an interval of 20 milliseconds and S is the number of samples by which the bandpass filter shifts the input speech samples. The second value, $R^i(1)$, provides a measure of the sinusoidal energy in the frequency band:

$$R^i(1) = \left| \sum_{n=0}^{N-1} s^i(n+S) s^{*i}(n) \right|^2.$$

Combination block 18 produces voiced/unvoiced parameters V^0 to V^K by selecting the minimum of a preliminary V/UV parameter from the first set and a function of a preliminary V/UV parameter from the second set. In particular, combination block produces the voiced/unvoiced parameters as:

$$V^k = \min(A^k, f_B(B^k))$$

where

$$f_B(B^k) = B^k + \alpha(k) \beta(\omega_o),$$

$$\beta(\omega_o) = 1.0, \text{ when } \omega_o \geq 2\pi/60.0, \text{ or}$$

$$2\pi/(60\omega_o), \text{ when } \omega_o < 2\pi/60.0$$

and $\alpha(k)$ is an increasing function of k . Because a preliminary V/UV parameter having a value close to zero has a higher probability of being correct than a preliminary V/UV parameter having a larger value, the selection of the minimum value results in the selection of the value that is most likely to be correct.

With reference to Fig. 6, in another embodiment, a first parameter estimator 14' produces the first preliminary V/UV estimate using an autocorrelation domain approach. Channel processing units 44 in first parameter estimator 14' divide speech signal $s(n)$ into at least two frequency bands and process the frequency bands to produce a first set of

frequency band signals, designated as $T^0(l) \dots T^K(l)$. There are eight channel processing units (K equals 7) and no remapping unit is necessary.

Next, voiced/unvoiced (V/UV) parameter estimation units 46, each associated with a channel processing unit 44, produce preliminary V/UV parameters A^0 to A^K by computing a ratio of the voiced energy in the frequency band at an estimated pitch period n_0 to the total energy in the frequency band and subtracting this ratio from 1:

$$A^k = 1.0 - E_v^k(n_0) / E_t^k.$$

The voiced energy in the frequency band is computed as:

$$E_v^k(n_0) = C(n_0) T^k(n_0)$$

where

$$C(n_0) = \frac{1}{\sum_{n=0}^{N-1} w(n) w(n+n_0)}$$

N is the number of samples in the window and typically has a value of 101, and $C(n_0)$ compensates for the window roll-off as a function of increasing autocorrelation lag. For non-integer values of n_0 , the voiced energy at the nearest three values of n are used with a parabolic interpolation method to obtain the voiced energy for n_0 . The total energy is determined as the voiced energy for n_0 equal to zero.

With reference to Fig. 7, when speech signal $s(n)$ enters a channel processing unit 44, components $s^i(n)$ belonging to a particular frequency band are isolated by a bandpass filter 48. Bandpass filter 48 uses downsampling to reduce computational requirements, and does so without any significant impact on system performance. Bandpass filter 48 can be implemented as a Finite Impulse Response (FIR) or Infinite Impulse Response (IIR) filter, or by using an FFT. A downsampling factor of S is achieved by shifting the input speech samples by S each time the filter outputs are computed.

A nonlinear operation unit 50 then performs a nonlinear operation on the isolated frequency band $s^i(n)$ to emphasize the fundamental frequency of the isolated frequency band $s^i(n)$. For complex values of $s^i(n)$ (i greater than zero), the absolute value, $|s^i(n)|$, is used. For the real value of $s^0(n)$, no nonlinear operation is performed.

The output of nonlinear operation unit 50 is passed through a highpass filter 52, and the output of the highpass filter is passed through an autocorrelation unit 54. A 101 point window is used, and, to reduce computation, the autocorrelation is only computed at a few samples nearest the pitch period.

With reference again to Fig. 4, second parameter estimator 16 may also use other approaches to produce the second voiced/unvoiced estimate. For example, well-known techniques such as using the height of the peak of the cepstrum, using the height of the peak of the autocorrelation of a linear prediction coder residual, MBE model parameter estimation methods, or IMBE (TM) model parameter estimation methods may be used. In addition, with reference again to Fig. 5, window and correlate unit 42 may produce autocorrelation values for the isolated frequency band $s^i(n)$ as:

$$R^i(l) = \text{Re} \left[\sum_n s^i(n+l) w(n+l) s^{*i}(n) w(n) \right]$$

where $w(n)$ is the window. With this approach, combination block 18 produces the voiced/unvoiced parameters as:

$$V^k = \min(A^k, B^k).$$

The fundamental frequency may be estimated using a number of approaches. First, with reference to Fig. 8, a fundamental frequency estimation unit 56 includes a combining unit 58 and an estimator 60. Combining unit 58 sums the $T^i(\omega)$ outputs of channel processing units 20 (Fig. 2) to produce $X(\omega)$. In an alternative approach, combining unit 58 could estimate a signal-to-noise ratio (SNR) for the output of each channel processing unit 20 and weigh the various

outputs so that an output with a higher SNR contributes more to $X(\omega)$ than does an output with a lower SNR.

Estimator 60 then estimates the fundamental frequency (ω_0) by selecting a value for ω_0 that maximizes $X(\omega_0)$ over an interval from $\omega_{\min}$ to $\omega_{\max}$. Since $X(\omega)$ is only available at discrete samples of ω , parabolic interpolation of $X(\omega_0)$ near ω_0 is used to improve accuracy of the estimate. Estimator 60 further improves the accuracy of the fundamental estimate by combining parabolic estimates near the peaks of the N harmonics of ω_0 within the bandwidth of $X(\omega)$.

Once an estimate of the fundamental frequency is determined, the voiced energy $E^v(\omega_0)$ is computed as:

$$E^v(\omega_0) = \sum_{n=1}^N \sum_{\omega_m \in I_n} X(\omega_m)$$

where

$$I_n = [(n-0.25)\omega_0, (n+0.25)\omega_0].$$

Thereafter, the voiced energy $E^v(0.5\omega_0)$ is computed and compared to $E^v(\omega_0)$ to select between ω_0 and $0.5\omega_0$ as the final estimate of the fundamental frequency.

With reference to Fig. 9, an alternative fundamental frequency estimation unit 62 includes a nonlinear operation unit 64, a windowing and Fast Fourier Transform (FFT) unit 66, and an estimator 68. Nonlinear operation unit 64 performs a nonlinear operation, the absolute value squared, on $s(n)$ to emphasize the fundamental frequency of $s(n)$ and to facilitate determination of the voiced energy when estimating ω_0 .

Windowing and FFT unit 66 multiplies the output of nonlinear operation unit 64 to segment it and computes an FFT, $X(\omega)$, of the resulting product. Finally, estimator 68, which works identically to estimator 60, generates an estimate of the fundamental frequency.

With reference to Fig. 10, a hybrid fundamental frequency estimation unit 70 includes a band combination and estimation unit 72, an IMBE estimation unit 74 and an estimate combination unit 76. Band combination and estimation unit 70 combines the outputs of channel processing units 20 (Fig. 2) using simple summation or a signal-to-noise ratio (SNR) weighting where bands with higher SNRs are given higher weight in the combination. From the combined signal ($U(\omega)$), unit 72 estimates a fundamental frequency and a probability that the fundamental frequency is correct. Unit 72 estimates the fundamental frequency by choosing the frequency that maximizes the voiced energy ($E_v(\omega_0)$) from the combined signal, which is determined as:

$$E_v(\omega_0) = \sum_{n=1}^N \sum_{\omega_m \in I_n} U(\omega_m)$$

where

$$I_n = [(n-0.25)\omega_0, (n+0.25)\omega_0].$$

and N is the number of harmonics of the fundamental frequency. The probability that ω_0 is correct is estimated by comparing $E_v(\omega_0)$ to the total energy E_t , which is computed as:

$$E_t = \sum_{\forall \omega_m > 0.5\omega_0} U(\omega_m).$$

When $E_v(\omega_0)$ is close to E_t , the probability estimate is near one. When $E_v(\omega_0)$ is close to one half of E_t , the probability estimate is near zero.

IMBE estimation unit 74 uses the well known IMBE technique, or a similar technique, to produce a second fundamental frequency estimate and probability of correctness. Thereafter, estimate combination unit 76 combines the two

fundamental frequency estimates to produce the final fundamental frequency estimate. The probabilities of correctness are used so that the estimate with higher probability of correctness is selected or given the most weight.

With reference to Fig. 11, a voiced/unvoiced parameter smoothing unit 78 performs a smoothing operation to remove voicing errors that might result from rapid transitions in the speech signal. Unit 78 produces a smoothed voiced/unvoiced parameter as:

$$v_s^k(n) = 1.0, \text{ when } v^k(n-1)v^k(n+1) = 1 \text{ and} \\ v^k(n), \text{ otherwise}$$

where the voiced/unvoiced parameters equal zero for unvoiced speech and one for voiced speech. When the voiced/unvoiced parameters have continuous values, with a value near zero corresponding to highly voiced speech, unit 78 produces a smoothed voiced/unvoiced parameter that is smoothed in both the time and frequency domains:

$$v_s^k(n) = \lambda^k(n) \min(v^k(n), \alpha^k(n), \beta^k(n), \gamma^k(n))$$

where

$$\alpha^k(n) = 2v^{k+1}(n), \text{ when } k=0, 1, \dots, K-1, \text{ or} \\ \infty, \text{ when } k=K;$$

$$\beta^k(n) = 2v^{k-1}(n), \text{ when } k=2, 3, \dots, K, \text{ or} \\ \infty, \text{ when } k=0, 1;$$

$$\gamma^k(n) = 0.25v^{k-1}(n) + 0.5v^k(n) + 0.25v^{k+1}(n), \\ \text{when } k=1, 2, \dots, K-1, \text{ or} \\ \infty, \text{ when } k=0, K;$$

$$\lambda^k(n) = 0.8, \text{ when } v_s^k(n-1) < T^k(n-1) \text{ and} \\ |\omega_o(n) - \omega_o(n-1)| < 0.25 |\omega_o(n)|, \text{ or} \\ 1, \text{ otherwise;}$$

and $T^k(n)$ is a threshold value that is a function of time and frequency.

With reference to Fig. 12, a voiced/unvoiced parameter improvement unit 80 produces improved voiced/unvoiced parameters by comparing the voiced/unvoiced parameter produced when the estimated fundamental frequency equals ω_o to a voiced/unvoiced parameter produced when the estimated fundamental frequency equals one half of ω_o and selecting the parameter having the lowest value. In particular, voiced/unvoiced parameter improvement unit 80 produces improved voiced/unvoiced parameters as:

$$A^k(\omega_o) = \min(A^k(\omega_o), A^k(0.5\omega_o))$$

where

$$A^k(\omega) = 1.0 - E_v^k(\omega)/E_t^k.$$

With reference to Fig. 13, an improved estimate of the fundamental frequency (ω_0) is generated according to a procedure 100. The initial fundamental frequency estimate ($\tilde{\omega}_0$) is generated according to one of the procedures described above and is used in step 101 to generate a set of evaluation frequencies $\tilde{\omega}^k$. The evaluation frequencies are typically chosen to be near the integer submultiples and multiples of $\tilde{\omega}_0$. Thereafter, functions are evaluated at this set of evaluation frequencies (step 102). The functions that are evaluated typically consist of the voiced energy function $E_v(\tilde{\omega}^k)$ and the normalized frame error $E_f(\tilde{\omega}^k)$. The normalized frame error is computed as

$$E_f(\tilde{\omega}^k) = 1.0 - E_v(\tilde{\omega}^k) / E_r(\tilde{\omega}^k).$$

The final fundamental frequency estimate is then selected (step 103) using the evaluation frequencies, the function values at the evaluation frequencies, the predicted fundamental frequency (described below), the final fundamental frequency estimates from previous frames, and the above function values from previous frames. When these inputs indicate that one evaluation frequency has a much higher probability of being the correct fundamental frequency than the others, then it is chosen. Otherwise, if two evaluation frequencies have similar probability of being correct and the normalized error for the previous frame is relatively low, then the evaluation frequency closest to the final fundamental frequency from the previous frame is chosen. Otherwise, if two evaluation frequencies have similar probability of being correct, then the one closest to the predicted fundamental frequency is chosen. The predicted fundamental frequency for the next frame is generated (step 104) using the final fundamental frequency estimates from the current and previous frames, a delta fundamental frequency, and normalized frame errors computed at the final fundamental frequency estimate for the current frame and previous frames. The delta fundamental frequency is computed from the frame to frame difference in the final fundamental frequency estimate when the normalized frame errors for these frames are relatively low and the percentage change in fundamental frequency is low, otherwise, it is computed from previous values. When the normalized error for the current frame is relatively low, the predicted fundamental for the current frame is set to the final fundamental frequency. The predicted fundamental for the next frame is set to the sum of the predicted fundamental for the current frame and the delta fundamental frequency for the current frame.

Claims

1. A method of analysing a digitized speech signal to determine excitation parameters for the digitized speech signal, preferably as a step in encoding speech, the method comprising dividing the digitized speech signal into one or more frequency band signals; and, preferably at regular intervals of time, performing the further step of: determining a first preliminary excitation parameter using a first method that includes performing a nonlinear operation on at least one of the frequency band signals to produce at least one modified frequency band signal and determining the first preliminary excitation parameter using the at least one modified frequency band signal; determining at least a second preliminary excitation parameter using at least a second method different from the said first method; and using the first and at least a second preliminary excitation parameters to determine an excitation parameter for the digitized speech signal.
2. A method according to Claim 1, wherein at least one of the second methods uses at least one of the frequency band signals without performing the said nonlinear operation.
3. A method according to Claims 1 or 2, wherein the excitation parameter comprises a voiced/unvoiced parameter for at least one frequency band, said parameter preferably having values that vary over a continuous range.
4. A method according to any preceding claim, further comprising determining a fundamental frequency for the digitized speech signal.
5. A method according to Claim 3, wherein the first preliminary excitation parameter comprises a first voiced/unvoiced parameter for the at least one modified frequency band signal, and wherein the first determining step includes determining the first voiced/unvoiced parameter by comparing voiced energy in the modified frequency band signal to total energy in the modified frequency band signal.
6. A method according to Claim 5, wherein the voiced energy in the modified frequency band signal corresponds to the energy associated with an estimated fundamental frequency for the digitized speech signal.



7. A method according to Claim 5, wherein the voiced energy in the modified frequency band signal corresponds to the energy associated with an estimated pitch period for the digitized speech signal.
- 5 8. A method according to Claim 5, wherein the second preliminary excitation parameter includes a second voiced/unvoiced parameter for the at least one frequency band signal, and wherein the second determining step includes determining the second voiced/unvoiced parameter by comparing sinusoidal energy in the at least one frequency band signal to total energy in the at least one frequency band signal.
- 10 9. A method according to Claim 5, wherein the second preliminary excitation parameter includes a second voiced/unvoiced parameter for the at least one frequency band signal, and wherein the second determining step includes determining the second voiced/unvoiced parameter by autocorrelating the at least one frequency band signal.
- 15 10. A method according to any preceding claim, wherein the said using step emphasizes the first preliminary excitation parameter over the second preliminary excitation parameter in determining the excitation parameter for the digitized speech signal when the first preliminary excitation parameter has a higher probability of being correct than does the second preliminary excitation parameter.
- 20 11. A method according to any preceding claim, further comprising smoothing the excitation parameter to produce a smoothed excitation parameter.
- 25 12. A method of analysing a digitized speech signal to determine excitation parameters for the digitized speech signal, preferably as a step in encoding speech, the method comprising the steps of: determining preliminary excitation parameters from the digitized speech signal; and smoothing the preliminary excitation parameters to produce excitation parameters.
- 30 13. A method according to Claim 12, wherein the preliminary excitation parameters include a preliminary voiced/unvoiced parameter for at least one frequency band and the excitation parameters include a voiced/unvoiced parameter for at least one frequency band, which voiced/unvoiced parameter preferably has values that vary over a continuous range.
- 35 14. A method according to Claim 13, wherein the excitation parameters include a fundamental frequency.
15. A method according to Claims 13 or 14, wherein the smoothing step makes the voiced/unvoiced parameter more voiced than the preliminary voiced/unvoiced parameter when voiced/unvoiced parameters that are nearby in time and/or frequency are voiced.
- 40 16. A method according to Claim 12, wherein the smoothing step is performed as a function of time and/or frequency.
- 45 17. A method of analysing a digitized speech signal to determine excitation parameters for the digitized speech signal, preferably as a step in encoding speech, the method comprising the steps of: estimating a fundamental frequency for the digitized speech signal; evaluating a voiced/unvoiced function using the estimated fundamental frequency to produce a first preliminary voiced/unvoiced parameter; evaluating the voiced/unvoiced function at least using one other frequency derived from the estimated fundamental frequency to produce at least one other preliminary voiced/unvoiced parameter; and combining the first and at least one other preliminary voiced/unvoiced parameters to produce a voiced/unvoiced parameter.
- 50 18. A method according to Claim 17, wherein the said at least one other frequency is derived from the said estimated fundamental frequency as a multiple or submultiple of the said estimated fundamental frequency.
- 55 19. A method according to Claim 17, wherein the combining step includes choosing the first preliminary voiced/unvoiced parameter as the voiced/unvoiced parameter when the first preliminary voiced/unvoiced parameter indicates that the digitized speech signal is more voiced than does the second preliminary voiced/unvoiced parameter.
20. A method of synthesizing speech using excitation parameters, where the excitation parameters are estimated by using a method for determining such parameters according to any preceding claim.
21. A method of analysing a digitized speech signal to determine a fundamental frequency estimate for the digitized speech signal, comprising the steps of: determining a predicted fundamental frequency estimate from previous

fundamental frequency estimates; determining an initial fundamental frequency estimate; evaluating an error function at the initial fundamental frequency estimate to produce a first error function value; evaluating the error function at at least one other frequency derived from the initial fundamental frequency estimate to produce at least one other error function value; selecting a fundamental frequency estimate using the predicted fundamental frequency estimate, the initial fundamental frequency estimate, the first error function value, and the at least one other error function value.

22. A method according to Claim 21, wherein the said at least one other frequency is derived from the said estimated fundamental frequency as a multiple or submultiple of the said estimated fundamental frequency.

23. A method according to Claim 21, wherein the predicted fundamental frequency is determined by adding a delta factor to a previous predicted fundamental frequency, which delta factor is preferably determined from previous first and at least one other error function values, the previous predicted fundamental frequency, and a previous delta factor.

24. A method of synthesizing speech using a fundamental frequency, where the fundamental frequency is estimated using a method according to any of Claims 21, 22 or 23.

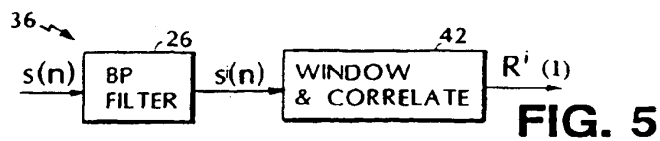
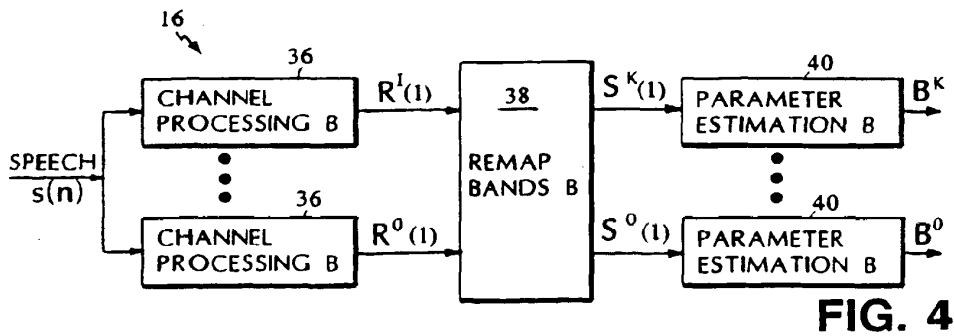
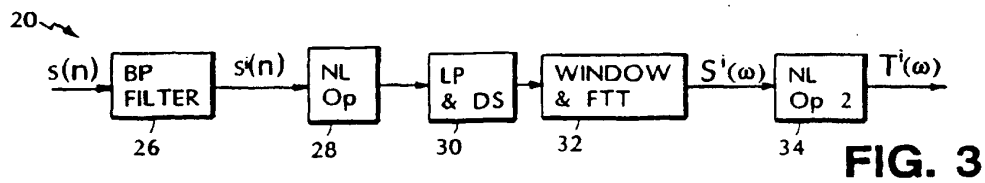
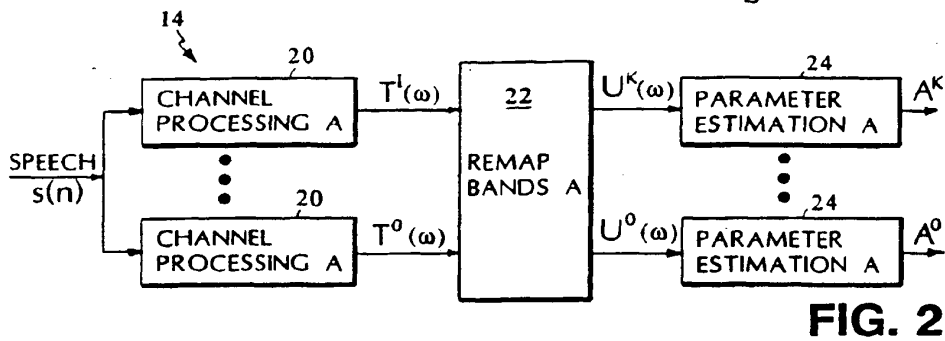
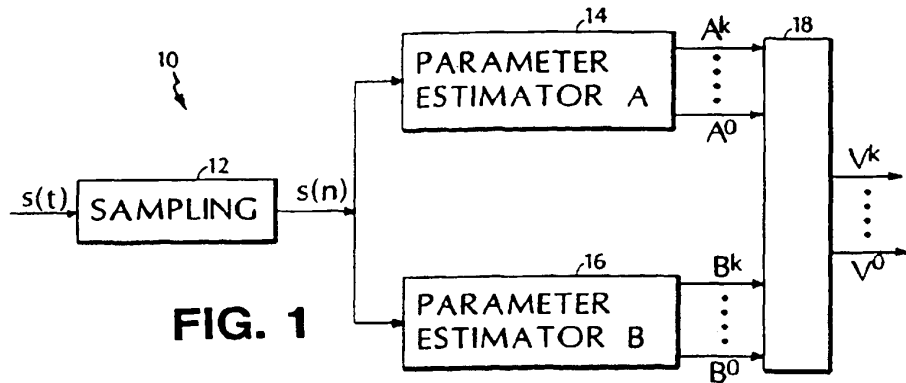
25. A system for analysing a digitized speech signal to determine excitation parameters for the digitized speech signal, comprising: means for dividing the digitized speech signal into one or more frequency band signals; means for determining a first preliminary excitation parameter using a first method that includes performing a nonlinear operation on at least one of the frequency band signals to produce at least one modified frequency band signal and determining the first preliminary excitation parameter using the at least one modified frequency band signal; means for determining a second preliminary excitation parameter using a second method that is different from the above said first method; and means for using the first and second preliminary excitation parameters to determine an excitation parameter for the digitized speech signal.

26. A system for analysing a digitized speech signal to determine excitation parameters for the digitized speech signal, comprising: means for determining preliminary excitation parameters from the digitized speech signal; and means for smoothing the preliminary excitation parameters to produce excitation parameters.

27. A system for analysing a digitized speech signal to determine modified excitation parameters for the digitized speech signal, comprising: means for estimating a fundamental frequency for the digitized speech signal; means for evaluating a voiced/unvoiced function using the estimated fundamental frequency to produce a first preliminary voiced/unvoiced parameter; means for evaluating the voiced/unvoiced function using another frequency derived from the estimated fundamental frequency to produce a second preliminary voiced/unvoiced parameter; and means for combining the first and second preliminary voiced/unvoiced parameters to produce a voiced/unvoiced parameter.

28. A system for analysing a digitized speech signal to determine a fundamental frequency estimate for the digitized speech signal, comprising: means for determining a predicted fundamental frequency estimate from previous fundamental frequency estimates; means for determining an initial fundamental frequency estimate; means for evaluating an error function at the initial fundamental frequency estimate to produce a first error function value; means for evaluating the error function at at least one other frequency derived from the initial fundamental frequency estimate to produce a second error function value; and means for selecting a fundamental frequency estimate using the predicted fundamental frequency estimate, the initial fundamental frequency estimate, the first error function value, and the second error function value.

29. A method of analysing a digitized speech signal to determine a voiced/unvoiced function for the digitized speech signal, comprising: dividing the digitized speech signal into at least two frequency band signals; determining a first preliminary voiced/unvoiced function for at least two of the frequency band signals using a first method; determining a second preliminary voiced/unvoiced function for at least two of the frequency band signals using a second method which is different from the above said first method; and using the first and second preliminary excitation parameters to determine a voiced/unvoiced function for at least two of the frequency band signals.



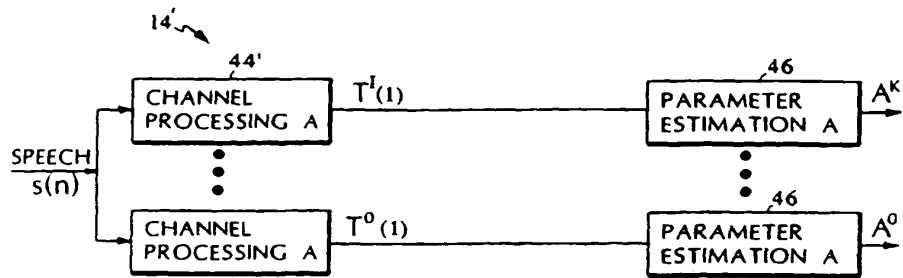


FIG. 6

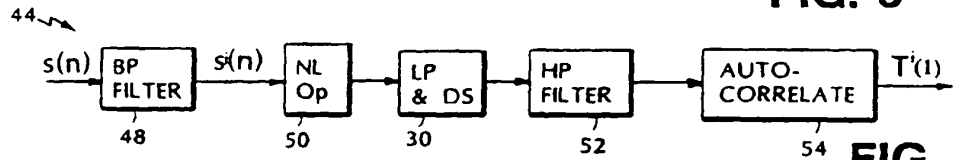


FIG. 7

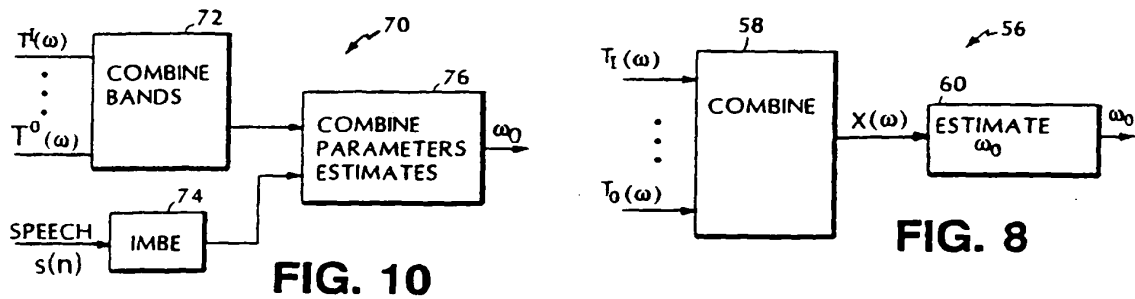


FIG. 8

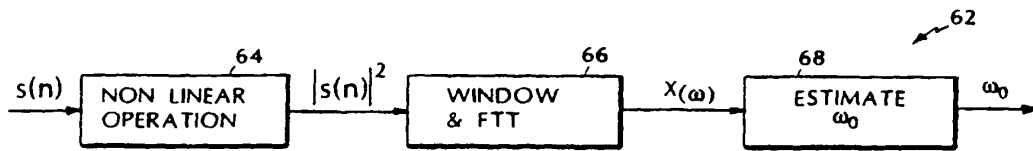


FIG. 9

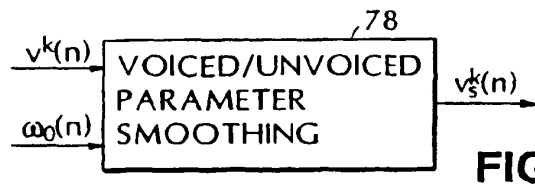


FIG. 11

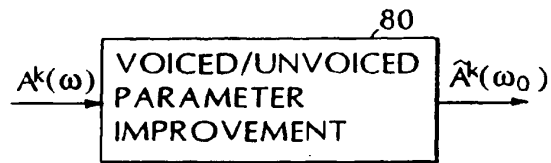


FIG. 12

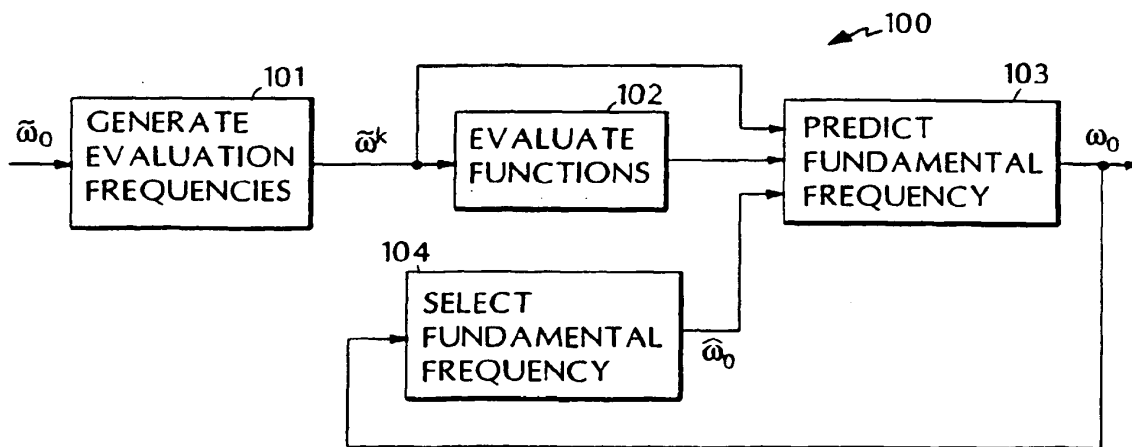


FIG. 13



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(54) Estimation of excitation parameters

(57) Excitation parameters for a digitized speech signal are determined by analysing the digitized speech signal. The digitized speech signal is divided into at least two frequency bands. A first preliminary excitation parameter is determined by performing a nonlinear operation on at least one of the frequency band signals to produce a modified frequency band signal and determining the first preliminary excitation parameter using the modified frequency band signal. A second preliminary

excitation parameter is determined using a method different from the first method. The first and second preliminary excitation parameters are used to determine an excitation parameter for the digitized speech signal. The method is useful in encoding speech. Speech synthesized using the parameters estimated based on the invention generates high quality speech at various bit rates useful for applications such as satellite voice communication.

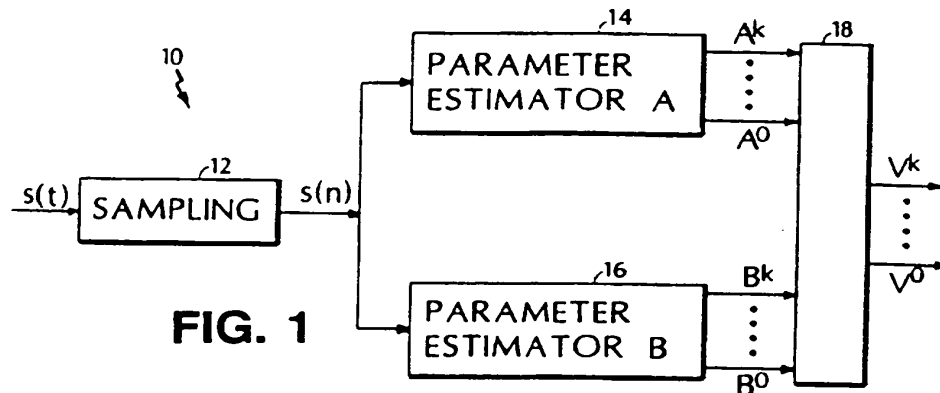


FIG. 1

EP 0 722 165 A2 CORRIGENDUM issued on 20.11.1996 (bibliography updates included)

Description

The invention has arisen from work seeking to improve the accuracy with which excitation parameters are estimated in speech analysis and synthesis.

Speech analysis and synthesis are widely used in applications such as telecommunications and voice recognition. A vocoder, which is a type of speech analysis/synthesis system, models speech as the response of a system to excitation over short time intervals. Examples of vocoder systems include linear prediction vocoders, homomorphic vocoders, channel vocoders, sinusoidal transform coders ("STC"), multiband excitation ("MBE") vocoders, improved multiband excitation ("IMBE (TM)") vocoders.

Vocoders typically synthesize speech based on excitation parameters and system parameters. Typically, an input signal is segmented using, for example, a Hamming window. Then, for each segment, system parameters and excitation parameters are determined. System parameters include the spectral envelope or the impulse response of the system. Excitation parameters include a fundamental frequency (or pitch) and a voiced/unvoiced parameter that indicates whether the input signal has pitch (or indicates the degree to which the input signal has pitch). In vocoders that divide the speech into frequency bands, such as IMBE (TM) vocoders, the excitation parameters may also include a voiced/unvoiced parameter for each frequency band rather than a single voiced/unvoiced parameter. Accurate excitation parameters are essential for high quality speech synthesis.

When the voiced/unvoiced parameters include only a single voiced/unvoiced decision for the entire frequency band, the synthesized speech tends to have a "buzzy" quality especially noticeable in regions of speech which contain mixed voicing or in voiced regions of noisy speech. A number of mixed excitation models have been proposed as potential solutions to the problem of "buzziness" in vocoders. In these models, periodic and noise-like excitations are mixed which have either time-invariant or time-varying spectral shapes.

In excitation models having time-invariant spectral shapes, the excitation signal consists of the sum of a periodic source and a noise source with fixed spectral envelopes. The mixture ratio controls the relative amplitudes of the periodic and noise sources. Examples of such models include Itakura and Saito, "Analysis Synthesis Telephony Based upon the Maximum Likelihood Method," Reports of 6th Int. Cong. Acoust., Tokyo, Japan, Paper C-5-5, pp. C17-20, 1968; and Kwon and Goldberg, "An Enhanced LPC Vocoder with No Voiced/Unvoiced Switch," IEEE Trans. on Acoust., Speech, and Signal Processing, vol. ASSP-32, no. 4, pp. 851-858, August 1984. In these excitation models a white noise source is added to a white periodic source. The mixture ratio between these sources is estimated from the height of the peak of the autocorrelation of the LPC residual.

In excitation models having time-varying spectral shapes, the excitation signal consists of the sum of a periodic source and a noise source with time varying spectral envelope shapes. Examples of such models include Fujimara, "An Approximation to Voice Aperiodicity," IEEE Trans. Audio and Electroacoust., pp. 68-72, March 1968; Makhoul et al., "A Mixed-Source Excitation Model for Speech Compression and Synthesis," IEEE Int. Conf. on Acoust. Sp. & Sig. Proc., April 1978, pp. 163-166; Kwon and Goldberg, "An Enhanced LPC Vocoder with No Voiced/Unvoiced Switch," IEEE Trans. on Acoust., Speech, and Signal Processing, vol. ASSP-32, no. 4, pp. 851-858, August 1984; and Griffin and Lim, "Multiband Excitation Vocoder," IEEE Trans. Acoust., Speech, Signal Processing, vol. ASSP-36, pp. 1223-1235, Aug. 1988.

In the excitation model proposed by Fujimara, the excitation spectrum is divided into three fixed frequency bands. A separate cepstral analysis is performed for each frequency band and a voiced/unvoiced decision for each frequency band is made based on the height of the cepstrum peak as a measure of periodicity.

In the excitation model proposed by Makhoul et al., the excitation signal consists of the sum of a low-pass periodic source and a high-pass noise source. The low-pass periodic source is generated by filtering a white pulse source with a variable cut-off low-pass filter. Similarly, the high-pass noise source was generated by filtering a white noise source with a variable cut-off high-pass filter. The cut-off frequencies for the two filters are equal and are estimated by choosing the highest frequency at which the spectrum is periodic. Periodicity of the spectrum is determined by examining the separation between consecutive peaks and determining whether the separations are the same, within some tolerance level.

In a second excitation model implemented by Kwon and Goldberg, a pulse source is passed through a variable gain low-pass filter and added to itself, and a white noise source is passed through a variable gain high-pass filter and added to itself. The excitation signal is the sum of the resultant pulse and noise sources with the relative amplitudes controlled by a voiced/unvoiced mixture ratio. The filter gains and voiced/unvoiced mixture ratio are estimated from the LPC residual signal with the constraint that the spectral envelope of the resultant excitation signal is flat.

In the multiband excitation model proposed by Griffin and Lim, a frequency dependent voiced/unvoiced mixture function is proposed. This model is restricted to a frequency dependent binary voiced/unvoiced decision for coding purposes. A further restriction of this model divides the spectrum into a finite number of frequency bands with a binary voiced/unvoiced decision for each band. The voiced/unvoiced information is estimated by comparing the speech spectrum to the closest periodic spectrum. When the error is below a threshold, the band is marked voiced, otherwise, the

band is marked unvoiced.

Excitation parameters may also be used in applications, such as speech recognition, where no speech synthesis is required. Once again, the accuracy of the excitation parameters directly affects the performance of such a system.

In one aspect, generally, the invention features a hybrid excitation parameter estimation technique that produces two sets of excitation parameters for a speech signal using two different approaches and combines the two sets to produce a single set of excitation parameters. In a first approach, the technique applies a nonlinear operation to the speech signal to emphasize the fundamental frequency of the speech signal. In a second approach, we use a different method which may or may not include a nonlinear operation. While the first approach produces highly accurate excitation parameters under most conditions, the second approach produces more accurate parameters under certain conditions. By using both approaches and combining the resulting sets of excitation parameters to produce a single set, our technique produces accurate results under a wider range of conditions than are produced by either of the approaches individually.

In typical approaches to determining excitation parameters, an analog speech signal $s(t)$ is sampled to produce a speech signal $s(n)$. Speech signal $s(n)$ is then multiplied by a window $w(n)$ to produce a windowed signal $s_w(n)$ that is commonly referred to as a speech segment or a speech frame. A Fourier transform is then performed on windowed signal $s_w(n)$ to produce a frequency spectrum $S_w(\omega)$ from which the excitation parameters are determined.

When speech signal $s(n)$ is periodic with a fundamental frequency ω_0 or pitch period n_0 (where n_0 equals $2\pi/\omega_0$), the frequency spectrum of speech signal $s(n)$ should be a line spectrum with energy at ω_0 and harmonics thereof (integral multiples of ω_0). As expected, $S_w(\omega)$ has spectral peaks that are centered around ω_0 and its harmonics. However, due to the windowing operation, the spectral peaks include some width, where the width depends on the length and shape of window $w(n)$ and tends to decrease as the length of window $w(n)$ increases. This window-induced error reduces the accuracy of the excitation parameters. Thus, to decrease the width of the spectral peaks, and to thereby increase the accuracy of the excitation parameters, the length of window $w(n)$ should be made as long as possible.

The maximum useful length of window $w(n)$ is limited. Speech signals are not stationary signals, and instead have fundamental frequencies that change over time. To obtain meaningful excitation parameters, an analyzed speech segment must have a substantially unchanged fundamental frequency. Thus, the length of window $w(n)$ must be short enough to ensure that the fundamental frequency will not change significantly within the window.

In addition to limiting the maximum length of window $w(n)$, a changing fundamental frequency tends to broaden the spectral peaks. This broadening effect increases with increasing frequency. For example, if the fundamental frequency changes by $\Delta\omega_0$ during the window, the frequency of the m th harmonic, which has a frequency of $m\omega_0$, changes by $m\Delta\omega_0$ so that the spectral peak corresponding to $m\omega_0$ is broadened more than the spectral peak corresponding to ω_0 . This increased broadening of the higher harmonics reduces the effectiveness of higher harmonics in the estimation of the fundamental frequency and the generation of voiced/unvoiced parameters for high frequency bands.

By applying a nonlinear operation to the speech signal, the increased impact on higher harmonics of a changing fundamental frequency is reduced or eliminated, and higher harmonics perform better in estimation of the fundamental frequency and determination of voiced/unvoiced parameters. Suitable nonlinear operations map from complex (or real) to real values and produce outputs that are nondecreasing functions of the magnitudes of the complex (or real) values. Such operations include, for example, the absolute value, the absolute value squared, the absolute value raised to some other power, or the log of the absolute value.

Nonlinear operations tend to produce output signals having spectral peaks at the fundamental frequencies of their input signals. This is true even when an input signal does not have a spectral peak at the fundamental frequency. For example, if a bandpass filter that only passes frequencies in the range between the third and fifth harmonics of ω_0 is applied to a speech signal $s(n)$, the output of the bandpass filter, $x(n)$, will have spectral peaks at $3\omega_0$, $4\omega_0$ and $5\omega_0$.

Though $x(n)$ does not have a spectral peak at ω_0 , $|x(n)|^2$ will have such a peak. For a real signal $x(n)$, $|x(n)|^2$ is equivalent to $x^2(n)$. As is well known, the Fourier transform of $x^2(n)$ is the convolution of $X(\omega)$, the Fourier transform of $x(n)$, with $X(\omega)$:

$$\sum_{n=-\infty}^{\infty} x^2(n) e^{-j\omega n} = \frac{1}{2\pi} \int_{u=-\pi}^{\pi} X(\omega-u) X(u) du.$$

The convolution of $X(\omega)$ with $X(\omega)$ has spectral peaks at frequencies equal to the differences between the frequencies for which $X(\omega)$ has spectral peaks. The differences between the spectral peaks of a periodic signal are the fundamental frequency and its multiples. Thus, in the example in which $X(\omega)$ has spectral peaks at $3\omega_0$, $4\omega_0$ and $5\omega_0$, $X(\omega)$ convolved with $X(\omega)$ has a spectral peak at ω_0 ($4\omega_0-3\omega_0$, $5\omega_0-4\omega_0$). For a typical periodic signal, the spectral peak at the funda-

mental frequency is likely to be the most prominent.

The above discussion also applies to complex signals. For a complex signal $x(n)$, the Fourier transform of $|x(n)|^2$ is:

$$\sum_{n=-\infty}^{\infty} |x(n)|^2 e^{-j\omega n} = \frac{1}{2\pi} \int_{-\pi}^{\pi} X(\omega+u) X^*(u) du.$$

This is an autocorrelation of $X(\omega)$ with $X^*(\omega)$, and also has the property that spectral peaks separated by $n\omega_0$ produce peaks at $n\omega_0$.

Even though $|x(n)|$, $|x(n)|^a$ for some real "a", and $\log |x(n)|$ are not the same as $|x(n)|^2$, the discussion above for $|x(n)|^2$ applies approximately at the qualitative level. For example, for $|x(n)| = y(n)^{0.5}$, where $y(n) = |x(n)|^2$, a Taylor series expansion of $y(n)$ can be expressed as:

$$|x(n)| = \sum_{k=0}^{\infty} c_k y^k(n).$$

Because multiplication is associative, the Fourier transform of the signal $y^k(n)$ is $Y(\omega)$ convolved with the Fourier transform of $y^{k-1}(n)$. The behavior for nonlinear operations other than $|x(n)|^2$ can be derived from $|x(n)|^2$ by observing the behavior of multiple convolutions of $Y(\omega)$ with itself. If $Y(\omega)$ has peaks at $n\omega_0$, then multiple convolutions of $Y(\omega)$ with itself will also have peaks at $n\omega_0$.

As shown, nonlinear operations emphasize the fundamental frequency of a periodic signal, and are particularly useful when the periodic signal includes significant energy at higher harmonics. However, the presence of the nonlinearity can degrade performance in some cases. For example, performance may be degraded when speech signal $s(n)$ is divided into multiple bands $s^i(n)$ using bandpass filters, where $s^i(n)$ denotes the result of bandpass filtering using the i th bandpass filter. If a single harmonic of the fundamental frequency is present in the pass band of the i th filter, the output of the filter is:

$$s^i(n) = A_k e^{j(\omega_k n + \theta_k)}$$

where ω_k is the frequency, θ_k is the phase, and A_k is the amplitude of the harmonic. When a nonlinearity such as the absolute value is applied to $s^i(n)$ to produce a value $y^i(n)$, the result is:

$$y^i(n) = |s^i(n)| = |A_k|$$

so that the frequency information has been completely removed from the signal $y^i(n)$. Removal of this frequency information can reduce the accuracy of parameter estimates.

Our hybrid technique provides significantly improved parameter estimation performance in cases for which the nonlinearity reduces the accuracy of parameter estimates while maintaining the benefits of the nonlinearity in the remaining cases. As described above, the hybrid technique includes combining parameter estimates based on the signal after the nonlinearity has been applied ($y^i(n)$) with parameter estimates based on the signal before the nonlinearity is applied ($s^i(n)$ or $s(n)$). The two approaches produce parameter estimates along with an indication of the probability of correctness of these parameter estimates. The parameter estimates are then combined giving higher weight to estimates with a higher probability of being correct.

In another aspect, generally, the invention features the application of smoothing techniques to the voiced/unvoiced parameters. Voiced/unvoiced parameters can be binary or continuous functions of time and/or frequency. Because these parameters tend to be smooth functions in at least one direction (positive or negative) of time or frequency, the estimates of these parameters can benefit from appropriate application of smoothing techniques in time and/or frequency.

The invention also features an improved technique for estimating voiced/unvoiced parameters. In vocoders such as linear prediction vocoders, homomorphic vocoders, channel vocoders, sinusoidal transform coders, multiband excitation vocoders, and IMBE (TM) vocoders, a pitch period n (or equivalently a fundamental frequency) is selected.

Thereafter, a function $f(n)$ is then evaluated at the selected pitch period (or fundamental frequency) to estimate the i th voiced/unvoiced parameter. However, for some speech signals, evaluation of this function only at the selected pitch period will result in reduced accuracy of one or more voiced/unvoiced parameter estimates. This reduced accuracy may result from speech signals that are more periodic at a multiple of the pitch period than at the pitch period, and may be frequency dependent so that only certain portions of the spectrum are more periodic at a multiple of the pitch period. Consequently, the voiced/unvoiced parameter estimation accuracy can be improved by evaluating the function $f(n)$ at the pitch period n and at its multiples, and thereafter combining the results of these evaluations.

In another aspect, the invention features an improved technique for estimating the fundamental frequency or pitch period. When the fundamental frequency ω_0 (or pitch period n_0) is estimated, there may be some ambiguity as to whether ω_0 or a submultiple or multiple of ω_0 is the best choice for the fundamental frequency. Since the fundamental frequency tends to be a smooth function of time for voiced speech, predictions of the fundamental frequency based on past estimates can be used to resolve ambiguities and improve the fundamental frequency estimate.

Other features and advantages of the invention will be apparent from the following description of the preferred embodiments, given by way of example only.

In the drawings:

Fig. 1 is a block diagram of a system for determining whether frequency bands of a signal are voiced or unvoiced.

Fig. 2 is a block diagram of a parameter estimation unit of the system of Fig. 1.

Fig. 3 is a block diagram of a channel processing unit of the parameter estimation unit of Fig. 2.

Fig. 4 is a block diagram of a parameter estimation unit of the system of Fig. 1.

Fig. 5 is a block diagram of a channel processing unit of the parameter estimation unit of Fig. 4.

Fig. 6 is a block diagram of a parameter estimation unit of the system of Fig. 1.

Fig. 7 is a block diagram of a channel processing unit of the parameter estimation unit of Fig. 6.

Figs. 8-10 are block diagrams of systems for determining the fundamental frequency of a signal.

Fig. 11 is a block diagram of voiced/unvoiced parameter smoothing unit.

Fig. 12 is a block diagram of voiced/unvoiced parameter improvement unit.

Fig. 13 is a block diagram of a fundamental frequency improvement unit.

Figs. 1-13 show the structure of a system for estimating excitation parameters, the various blocks and units of which are preferably implemented with software.

With reference to Fig. 1, a voiced/unvoiced determination system 10 includes a sampling unit 12 that samples an analog speech signal $s(t)$ to produce a speech signal $s(n)$. For typical speech coding applications, the sampling rate ranges between six kilohertz and ten kilohertz.

Speech signal $s(n)$ is supplied to a first parameter estimator 14 that divides the speech signal into $k+1$ bands and produces a first set of preliminary voiced/unvoiced ("V/UV") parameters (A^0 to A^K) corresponding to a first estimate as to whether the signals in the bands are voiced or unvoiced. Speech signal $s(n)$ is also supplied to a second parameter estimator 16 that produces a second set of preliminary V/UV parameters (B^0 to B^K) that correspond to a second estimate as to whether the signals in the bands are voiced or unvoiced. The two sets of preliminary V/UV parameters are combined by a combination block 18 to produce a set of V/UV parameters (V^0 to V^K).

With reference to Fig. 2, first parameter estimator 14 produces the first voiced/unvoiced estimate using a frequency domain approach. Channel processing units 20 in first parameter estimator 14 divide speech signal $s(n)$ into at least two frequency bands and process the frequency bands to produce a first set of frequency band signals, designated as $T^0(\omega) \dots T^1(\omega)$. As discussed below, channel processing units 20 are differentiated by the parameters of a bandpass filter used in the first stage of each channel processing unit 20. In the described embodiment, there are sixteen channel processing units (I equals 15).

A remap unit 22 transforms the first set of frequency band signals to produce a second set of frequency band signals, designated as $U^0(\omega) \dots U^K(\omega)$. In the described embodiment, there are eight frequency band signals in the second set of frequency band signals (K equals 7). Thus, remap unit 22 maps the frequency band signals from the sixteen channel processing units 20 into eight frequency band signals. Remap unit 20 does so by combining consecutive pairs of frequency band signals from the first set into single frequency band signals in the second set. For example, $T^0(\omega)$ and $T^1(\omega)$ are combined to produce $U^0(\omega)$, and $T^{14}(\omega)$ and $T^{15}(\omega)$ are combined to produce $U^7(\omega)$. Other approaches to remapping could also be used.

Next, voiced/unvoiced parameter estimation units 24, each associated with a frequency band signal from the second set, produce preliminary V/UV parameters A^0 to A^K by computing a ratio of the voiced energy in the frequency band at an estimated fundamental frequency ω^0 to the total energy in the frequency band and subtracting this ratio from 1:

$$A^k = 1.0 - E_v^k(\omega_0) / E_t^k$$

The voiced energy in the frequency band is computed as:

$$E_V^k(\omega_o) = \sum_{n=1}^N \sum_{\omega_m \in I_n} U^k(\omega_m)$$

where

$$I_n = [(n-0.25) \omega_o, (n+0.25) \omega_o],$$

and N is the number of harmonics of the fundamental frequency ω_o being considered. V/UV parameter estimation units 24 determine the total energy of their associated frequency band signals as:

$$E_T^k = \sum_{\forall \omega_m > 0.5 \omega_o} U^k(\omega_m).$$

The degree to which the frequency band signal is voiced varies indirectly with the value of the preliminary V/UV parameter. Thus, the frequency band signal is highly voiced when the preliminary V/UV parameter is near zero and is highly unvoiced when the parameter is greater than or equal to one half.

With reference to Fig. 3, when speech signal $s(n)$ enters a channel processing unit 20, components $s^i(n)$ belonging to a particular frequency band are isolated by a bandpass filter 26. Bandpass filter 26 uses downsampling to reduce computational requirements, and does so without any significant impact on system performance. Bandpass filter 26 can be implemented as a Finite Impulse Response (FIR) or Infinite Impulse Response (IIR) filter, or by using an FFT. In the described embodiment, bandpass filter 26 is implemented using a thirty two point real input FFT to compute the outputs of a thirty two point FIR filter at seventeen frequencies, and achieves a downsampling factor of S by shifting the input by S samples each time the FFT is computed. For example, if a first FFT used samples one through thirty two, a downsampling factor of ten would be achieved by using samples eleven through forty two in a second FFT.

A first nonlinear operation unit 28 then performs a nonlinear operation on the isolated frequency band $s^i(n)$ to emphasize the fundamental frequency of the isolated frequency band $s^i(n)$. For complex values of $s^i(n)$ (i greater than zero), the absolute value, $|s^i(n)|$, is used. For the real value of $s^0(n)$, $s^0(n)$ is used if $s^0(n)$ is greater than zero and zero is used if $s^0(n)$ is less than or equal to zero.

The output of nonlinear operation unit 28 is passed through a lowpass filtering and downsampling unit 30 to reduce the data rate and consequently reduce the computational requirements of later components of the system. Lowpass filtering and downsampling unit 30 uses an FIR filter computed every other sample for a downsampling factor of two.

A windowing and FFT unit 32 multiplies the output of lowpass filtering and downsampling unit 30 by a window and computes a real input FFT, $S^i(\omega)$, of the product. Typically, windowing and FFT unit 32 uses a Hamming window and a real input FFT.

Finally, a second nonlinear operation unit 34 performs a nonlinear operation on $S^i(\omega)$ to facilitate estimation of voiced or total energy and to ensure that the outputs of channel processing units 20, $T^i(\omega)$, combine constructively if used in fundamental frequency estimation. The absolute value squared is used because it makes all components of $T^i(\omega)$ real and positive.

With reference to Fig. 4, second parameter estimator 16 produces the second preliminary V/UV estimates using a sinusoid detector/estimator. Channel processing units 36 in second parameter estimator 16 divide speech signal $s(n)$ into at least two frequency bands and process the frequency bands to produce a first set of signals, designated as $R^0(l) \dots R^l(l)$. Channel processing units 36 are differentiated by the parameters of a bandpass filter used in the first stage of each channel processing unit 36. In the described embodiment, there are sixteen channel processing units (l equals 15). The number of channels (value of l) in Fig. 4 does not have to equal the number of channels (value of l) in Fig. 2.

A remap unit 38 transforms the first set of signals to produce a second set of signals, designated as $S^0(l) \dots S^K(l)$. The remap unit can be an identity system. In the described embodiment, there are eight signals in the second set of signals (K equals 7). Thus, remap unit 38 maps the signals from the sixteen channel processing units 36 into eight signals. Remap unit 38 does so by combining consecutive pairs of signals from the first set into single signals in the

second set. For example, $R^0(l)$ and $R^1(l)$ are combined to produce $S^0(l)$, and $R^{14}(l)$ and $R^{15}(l)$ are combined to produce $S^7(l)$. Other approaches to remapping could also be used.

Next, V/UV parameter estimation units 40, each associated with a signal from the second set, produce preliminary V/UV parameters B^0 to B^K by computing a ratio of the sinusoidal energy in the signal to the total energy in the signal and subtracting this ratio from 1:

$$B^k = 1.0 - S^k(1)/S^k(0).$$

With reference to Fig. 5, when speech signal $s(n)$ enters a channel processing unit 36, components $s^i(n)$ belonging to a particular frequency band are isolated by a bandpass filter 26 that operates identically to the bandpass filters of channel processing units 20 (see Fig. 3). It should be noted that, to reduce computation requirements, the same bandpass filters may be used in channel processing units 20 and 36, with the outputs of each filter being supplied to a first nonlinear operation unit 28 of a channel processing unit 20 and a window and correlate unit 42 of a channel processing unit 36.

A window and correlate unit 42 then produces two correlation values for the isolated frequency band $s^i(n)$. The first value, $R^i(0)$, provides a measure of the total energy in the frequency band:

$$R^i(0) = \left[\frac{1}{2} \sum_{n=0}^{N-1} [|s^i(n)|^2 + |s^i(n+S)|^2] \right]^2$$

where N is related to the size of the window and typically defines an interval of 20 milliseconds and S is the number of samples by which the bandpass filter shifts the input speech samples. The second value, $R^i(1)$, provides a measure of the sinusoidal energy in the frequency band:

$$R^i(1) = \left| \sum_{n=0}^{N-1} s^i(n+S) s^{*i}(n) \right|^2.$$

Combination block 18 produces voiced/unvoiced parameters V^0 to V^K by selecting the minimum of a preliminary V/UV parameter from the first set and a function of a preliminary V/UV parameter from the second set. In particular, combination block produces the voiced/unvoiced parameters as:

$$V^k = \min(A^k, f_B(B^k))$$

where

$$f_B(B^k) = B^k + \alpha(k) \beta(\omega_o).$$

$$\beta(\omega_o) = 1.0, \text{ when } \omega_o \geq 2\pi/60.0, \text{ or}$$

$$2\pi/(60\omega_o), \text{ when } \omega_o < 2\pi/60.0$$

and $\alpha(k)$ is an increasing function of k . Because a preliminary V/UV parameter having a value close to zero has a higher probability of being correct than a preliminary V/UV parameter having a larger value, the selection of the minimum value results in the selection of the value that is most likely to be correct.

With reference to Fig. 6, in another embodiment, a first parameter estimator 14' produces the first preliminary V/UV estimate using an autocorrelation domain approach. Channel processing units 44 in first parameter estimator 14' divide speech signal $s(n)$ into at least two frequency bands and process the frequency bands to produce a first set of

frequency band signals, designated as $T^0(l)$.. $T^K(l)$. There are eight channel processing units (K equals 7) and no remapping unit is necessary.

Next, voiced/unvoiced (V/UV) parameter estimation units 46, each associated with a channel processing unit 44, produce preliminary V/UV parameters A^0 to A^K by computing a ratio of the voiced energy in the frequency band at an estimated pitch period n_0 to the total energy in the frequency band and subtracting this ratio from 1:

$$A^k = 1.0 - E_v^k(n_0) / E_t^k$$

The voiced energy in the frequency band is computed as:

$$E_v^k(n_0) = C(n_0) T^k(n_0)$$

where

$$C(n_0) = \frac{1}{\sum_{n=0}^{N-1} w(n) w(n+n_0)}$$

N is the number of samples in the window and typically has a value of 101, and $C(n_0)$ compensates for the window roll-off as a function of increasing autocorrelation lag. For non-integer values of n_0 , the voiced energy at the nearest three values of n are used with a parabolic interpolation method to obtain the voiced energy for n_0 . The total energy is determined as the voiced energy for n_0 equal to zero.

With reference to Fig. 7, when speech signal $s(n)$ enters a channel processing unit 44, components $s^i(n)$ belonging to a particular frequency band are isolated by a bandpass filter 48. Bandpass filter 48 uses downsampling to reduce computational requirements, and does so without any significant impact on system performance. Bandpass filter 48 can be implemented as a Finite Impulse Response (FIR) or Infinite Impulse Response (IIR) filter, or by using an FFT. A downsampling factor of S is achieved by shifting the input speech samples by S each time the filter outputs are computed.

A nonlinear operation unit 50 then performs a nonlinear operation on the isolated frequency band $s^i(n)$ to emphasize the fundamental frequency of the isolated frequency band $s^i(n)$. For complex values of $s^i(n)$ (i greater than zero), the absolute value, $|s^i(n)|$, is used. For the real value of $s^0(n)$, no nonlinear operation is performed.

The output of nonlinear operation unit 50 is passed through a highpass filter 52, and the output of the highpass filter is passed through an autocorrelation unit 54. A 101 point window is used, and, to reduce computation, the autocorrelation is only computed at a few samples nearest the pitch period.

With reference again to Fig. 4, second parameter estimator 16 may also use other approaches to produce the second voiced/unvoiced estimate. For example, well-known techniques such as using the height of the peak of the cepstrum, using the height of the peak of the autocorrelation of a linear prediction coder residual, MBE model parameter estimation methods, or IMBE (TM) model parameter estimation methods may be used. In addition, with reference again to Fig. 5, window and correlate unit 42 may produce autocorrelation values for the isolated frequency band $s^i(n)$ as:

$$R^i(l) = \text{Re} \left[\sum_n s^i(n+l) w(n+l) s^{*i}(n) w(n) \right]$$

where $w(n)$ is the window. With this approach, combination block 18 produces the voiced/unvoiced parameters as:

$$V^k = \min(A^k, B^k).$$

The fundamental frequency may be estimated using a number of approaches. First, with reference to Fig. 8, a fundamental frequency estimation unit 56 includes a combining unit 58 and an estimator 60. Combining unit 58 sums the $T^i(\omega)$ outputs of channel processing units 20 (Fig. 2) to produce $X(\omega)$. In an alternative approach, combining unit 58 could estimate a signal-to-noise ratio (SNR) for the output of each channel processing unit 20 and weigh the various

outputs so that an output with a higher SNR contributes more to $X(\omega)$ than does an output with a lower SNR.

Estimator 60 then estimates the fundamental frequency (ω_o) by selecting a value for ω_o that maximizes $X(\omega_o)$ over an interval from $\omega_{\min}$ to $\omega_{\max}$. Since $X(\omega)$ is only available at discrete samples of ω , parabolic interpolation of $X(\omega_o)$ near ω_o is used to improve accuracy of the estimate. Estimator 60 further improves the accuracy of the fundamental estimate by combining parabolic estimates near the peaks of the N harmonics of ω_o within the bandwidth of $X(\omega)$.

Once an estimate of the fundamental frequency is determined, the voiced energy $E^v(\omega_o)$ is computed as:

$$E^v(\omega_o) = \sum_{n=1}^N \sum_{\omega_m \in I_n} X(\omega_m)$$

where

$$I_n = [(n-0.25)\omega_o, (n+0.25)\omega_o].$$

Thereafter, the voiced energy $E^v(0.5\omega_o)$ is computed and compared to $E^v(\omega_o)$ to select between ω_o and $0.5\omega_o$ as the final estimate of the fundamental frequency.

With reference to Fig. 9, an alternative fundamental frequency estimation unit 62 includes a nonlinear operation unit 64, a windowing and Fast Fourier Transform (FFT) unit 66, and an estimator 68. Nonlinear operation unit 64 performs a nonlinear operation, the absolute value squared, on $s(n)$ to emphasize the fundamental frequency of $s(n)$ and to facilitate determination of the voiced energy when estimating ω_o .

Windowing and FFT unit 66 multiplies the output of nonlinear operation unit 64 to segment it and computes an FFT, $X(\omega)$, of the resulting product. Finally, estimator 68, which works identically to estimator 60, generates an estimate of the fundamental frequency.

With reference to Fig. 10, a hybrid fundamental frequency estimation unit 70 includes a band combination and estimation unit 72, an IMBE estimation unit 74 and an estimate combination unit 76. Band combination and estimation unit 70 combines the outputs of channel processing units 20 (Fig. 2) using simple summation or a signal-to-noise ratio (SNR) weighting where bands with higher SNRs are given higher weight in the combination. From the combined signal ($U(\omega)$), unit 72 estimates a fundamental frequency and a probability that the fundamental frequency is correct. Unit 72 estimates the fundamental frequency by choosing the frequency that maximizes the voiced energy ($E_v(\omega_o)$) from the combined signal, which is determined as:

$$E_v(\omega_o) = \sum_{n=1}^N \sum_{\omega_m \in I_n} U(\omega_m)$$

where

$$I_n = [(n-0.25)\omega_o, (n+0.25)\omega_o].$$

and N is the number of harmonics of the fundamental frequency. The probability that ω_o is correct is estimated by comparing $E_v(\omega_o)$ to the total energy E_t , which is computed as:

$$E_t = \sum_{\forall \omega_m > 0.5\omega_o} U(\omega_m).$$

When $E_v(\omega_o)$ is close to E_t , the probability estimate is near one. When $E_v(\omega_o)$ is close to one half of E_t , the probability estimate is near zero.

IMBE estimation unit 74 uses the well known IMBE technique, or a similar technique, to produce a second fundamental frequency estimate and probability of correctness. Thereafter, estimate combination unit 76 combines the two

fundamental frequency estimates to produce the final fundamental frequency estimate. The probabilities of correctness are used so that the estimate with higher probability of correctness is selected or given the most weight.

With reference to Fig. 11, a voiced/unvoiced parameter smoothing unit 78 performs a smoothing operation to remove voicing errors that might result from rapid transitions in the speech signal. Unit 78 produces a smoothed voiced/unvoiced parameter as:

$$v_s^k(n) = 1.0, \text{ when } v^k(n-1)v^k(n+1) = 1 \text{ and}$$

$$v^k(n), \text{ otherwise}$$

where the voiced/unvoiced parameters equal zero for unvoiced speech and one for voiced speech. When the voiced/unvoiced parameters have continuous values, with a value near zero corresponding to highly voiced speech, unit 78 produces a smoothed voiced/unvoiced parameter that is smoothed in both the time and frequency domains:

$$v_s^k(n) = \lambda^k(n) \min(v^k(n), \alpha^k(n), \beta^k(n), \gamma^k(n))$$

where

$$\alpha^k(n) = 2v^{k+1}(n), \text{ when } k=0, 1, \dots, K-1, \text{ or} \\ \infty, \text{ when } k=K;$$

$$\beta^k(n) = 2v^{k-1}(n), \text{ when } k=2, 3, \dots, K, \text{ or} \\ \infty, \text{ when } k=0, 1;$$

$$\gamma^k(n) = 0.25v^{k-1}(n) + 0.5v^k(n) + 0.25v^{k+1}(n), \\ \text{when } k=1, 2, \dots, K-1, \text{ or} \\ \infty, \text{ when } k=0, K;$$

$$\lambda^k(n) = 0.8, \text{ when } v_s^k(n-1) < T^k(n-1) \text{ and} \\ |\omega_o(n) - \omega_o(n-1)| < 0.25 |\omega_o(n)|, \text{ or} \\ 1, \text{ otherwise;}$$

and $T^k(n)$ is a threshold value that is a function of time and frequency.

With reference to Fig. 12, a voiced/unvoiced parameter improvement unit 80 produces improved voiced/unvoiced parameters by comparing the voiced/unvoiced parameter produced when the estimated fundamental frequency equals ω_o to a voiced/unvoiced parameter produced when the estimated fundamental frequency equals one half of ω_o and selecting the parameter having the lowest value. In particular, voiced/unvoiced parameter improvement unit 80 produces improved voiced/unvoiced parameters as:

$$A^k(\omega_o) = \min(A^k(\omega_o), A^k(0.5\omega_o))$$

where

$$A^k(\omega) = 1.0 - E_v^k(\omega)/E_t^k.$$

With reference to Fig. 13, an improved estimate of the fundamental frequency (ω_0) is generated according to a procedure 100. The initial fundamental frequency estimate ($\tilde{\omega}_0$) is generated according to one of the procedures described above and is used in step 101 to generate a set of evaluation frequencies $\tilde{\omega}^k$. The evaluation frequencies are typically chosen to be near the integer submultiples and multiples of $\tilde{\omega}_0$. Thereafter, functions are evaluated at this set of evaluation frequencies (step 102). The functions that are evaluated typically consist of the voiced energy function $E_v(\tilde{\omega}^k)$ and the normalized frame error $E_f(\tilde{\omega}^k)$. The normalized frame error is computed as

$$E_f(\tilde{\omega}^k) = 1.0 - E_v(\tilde{\omega}^k) / E_c(\tilde{\omega}^k) .$$

The final fundamental frequency estimate is then selected (step 103) using the evaluation frequencies, the function values at the evaluation frequencies, the predicted fundamental frequency (described below), the final fundamental frequency estimates from previous frames, and the above function values from previous frames. When these inputs indicate that one evaluation frequency has a much higher probability of being the correct fundamental frequency than the others, then it is chosen. Otherwise, if two evaluation frequencies have similar probability of being correct and the normalized error for the previous frame is relatively low, then the evaluation frequency closest to the final fundamental frequency from the previous frame is chosen. Otherwise, if two evaluation frequencies have similar probability of being correct, then the one closest to the predicted fundamental frequency is chosen. The predicted fundamental frequency for the next frame is generated (step 104) using the final fundamental frequency estimates from the current and previous frames, a delta fundamental frequency, and normalized frame errors computed at the final fundamental frequency estimate for the current frame and previous frames. The delta fundamental frequency is computed from the frame to frame difference in the final fundamental frequency estimate when the normalized frame errors for these frames are relatively low and the percentage change in fundamental frequency is low, otherwise, it is computed from previous values. When the normalized error for the current frame is relatively low, the predicted fundamental for the current frame is set to the final fundamental frequency. The predicted fundamental for the next frame is set to the sum of the predicted fundamental for the current frame and the delta fundamental frequency for the current frame.

Claims

1. A method of analysing a digitized speech signal to determine excitation parameters for the digitized speech signal, preferably as a step in encoding speech, the method comprising dividing the digitized speech signal into one or more frequency band signals; and, preferably at regular intervals of time, performing the further step of: determining a first preliminary excitation parameter using a first method that includes performing a nonlinear operation on at least one of the frequency band signals to produce at least one modified frequency band signal and determining the first preliminary excitation parameter using the at least one modified frequency band signal; determining at least a second preliminary excitation parameter using at least a second method different from the said first method; and using the first and at least a second preliminary excitation parameters to determine an excitation parameter for the digitized speech signal.
2. A method according to Claim 1, wherein at least one of the second methods uses at least one of the frequency band signals without performing the said nonlinear operation.
3. A method according to Claims 1 or 2, wherein the excitation parameter comprises a voiced/unvoiced parameter for at least one frequency band, said parameter preferably having values that vary over a continuous range.
4. A method according to any preceding claim, further comprising determining a fundamental frequency for the digitized speech signal.
5. A method according to Claim 3, wherein the first preliminary excitation parameter comprises a first voiced/unvoiced parameter for the at least one modified frequency band signal, and wherein the first determining step includes determining the first voiced/unvoiced parameter by comparing voiced energy in the modified frequency band signal to total energy in the modified frequency band signal.
6. A method according to Claim 5, wherein the voiced energy in the modified frequency band signal corresponds to the energy associated with an estimated fundamental frequency for the digitized speech signal.

7. A method according to Claim 5, wherein the voiced energy in the modified frequency band signal corresponds to the energy associated with an estimated pitch period for the digitized speech signal.
- 5 8. A method according to Claim 5, wherein the second preliminary excitation parameter includes a second voiced/unvoiced parameter for the at least one frequency band signal, and wherein the second determining step includes determining the second voiced/unvoiced parameter by comparing sinusoidal energy in the at least one frequency band signal to total energy in the at least one frequency band signal.
- 10 9. A method according to Claim 5, wherein the second preliminary excitation parameter includes a second voiced/unvoiced parameter for the at least one frequency band signal, and wherein the second determining step includes determining the second voiced/unvoiced parameter by autocorrelating the at least one frequency band signal.
- 15 10. A method according to any preceding claim, wherein the said using step emphasizes the first preliminary excitation parameter over the second preliminary excitation parameter in determining the excitation parameter for the digitized speech signal when the first preliminary excitation parameter has a higher probability of being correct than does the second preliminary excitation parameter.
- 20 11. A method according to any preceding claim, further comprising smoothing the excitation parameter to produce a smoothed excitation parameter.
- 25 12. A method of analysing a digitized speech signal to determine excitation parameters for the digitized speech signal, preferably as a step in encoding speech, the method comprising the steps of: determining preliminary excitation parameters from the digitized speech signal; and smoothing the preliminary excitation parameters to produce excitation parameters.
- 30 13. A method according to Claim 12, wherein the preliminary excitation parameters include a preliminary voiced/unvoiced parameter for at least one frequency band and the excitation parameters include a voiced/unvoiced parameter for at least one frequency band, which voiced/unvoiced parameter preferably has values that vary over a continuous range.
- 35 14. A method according to Claim 13, wherein the excitation parameters include a fundamental frequency.
15. A method according to Claims 13 or 14, wherein the smoothing step makes the voiced/unvoiced parameter more voiced than the preliminary voiced/unvoiced parameter when voiced/unvoiced parameters that are nearby in time and/or frequency are voiced.
16. A method according to Claim 12, wherein the smoothing step is performed as a function of time and/or frequency.
- 40 17. A method of analysing a digitized speech signal to determine excitation parameters for the digitized speech signal, preferably as a step in encoding speech, the method comprising the steps of: estimating a fundamental frequency for the digitized speech signal; evaluating a voiced/unvoiced function using the estimated fundamental frequency to produce a first preliminary voiced/unvoiced parameter; evaluating the voiced/unvoiced function at least using one other frequency derived from the estimated fundamental frequency to produce at least one other preliminary voiced/unvoiced parameter; and combining the first and at least one other preliminary voiced/unvoiced parameters to produce a voiced/unvoiced parameter.
- 45 18. A method according to Claim 17, wherein the said at least one other frequency is derived from the said estimated fundamental frequency as a multiple or submultiple of the said estimated fundamental frequency.
- 50 19. A method according to Claim 17, wherein the combining step includes choosing the first preliminary voiced/unvoiced parameter as the voiced/unvoiced parameter when the first preliminary voiced/unvoiced parameter indicates that the digitized speech signal is more voiced than does the second preliminary voiced/unvoiced parameter.
- 55 20. A method of synthesizing speech using excitation parameters, where the excitation parameters are estimated by using a method for determining such parameters according to any preceding claim.
21. A method of analysing a digitized speech signal to determine a fundamental frequency estimate for the digitized speech signal, comprising the steps of: determining a predicted fundamental frequency estimate from previous

fundamental frequency estimates; determining an initial fundamental frequency estimate; evaluating an error function at the initial fundamental frequency estimate to produce a first error function value; evaluating the error function at at least one other frequency derived from the initial fundamental frequency estimate to produce at least one other error function value; selecting a fundamental frequency estimate using the predicted fundamental frequency estimate, the initial fundamental frequency estimate, the first error function value, and the at least one other error function value.

22. A method according to Claim 21, wherein the said at least one other frequency is derived from the said estimated fundamental frequency as a multiple or submultiple of the said estimated fundamental frequency.

23. A method according to Claim 21, wherein the predicted fundamental frequency is determined by adding a delta factor to a previous predicted fundamental frequency, which delta factor is preferably determined from previous first and at least one other error function values, the previous predicted fundamental frequency, and a previous delta factor.

24. A method of synthesizing speech using a fundamental frequency, where the fundamental frequency is estimated using a method according to any of Claims 21, 22 or 23.

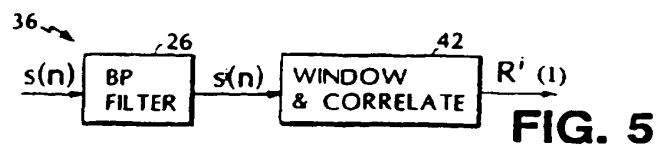
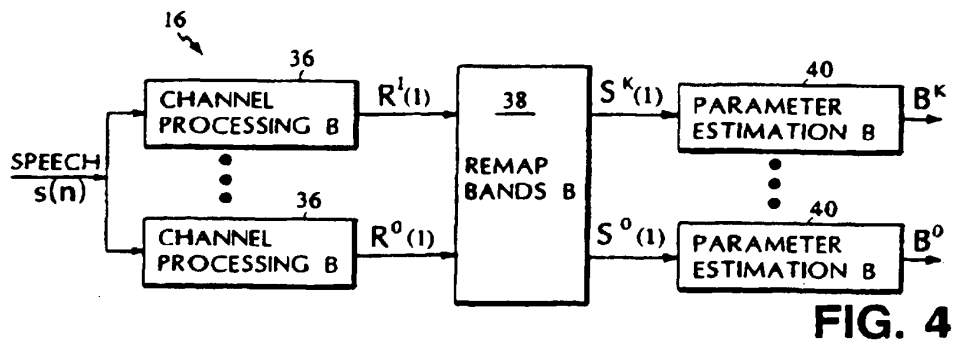
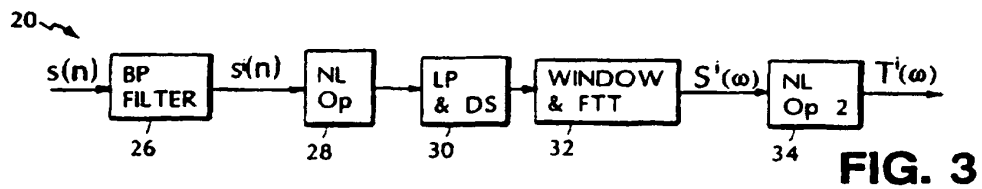
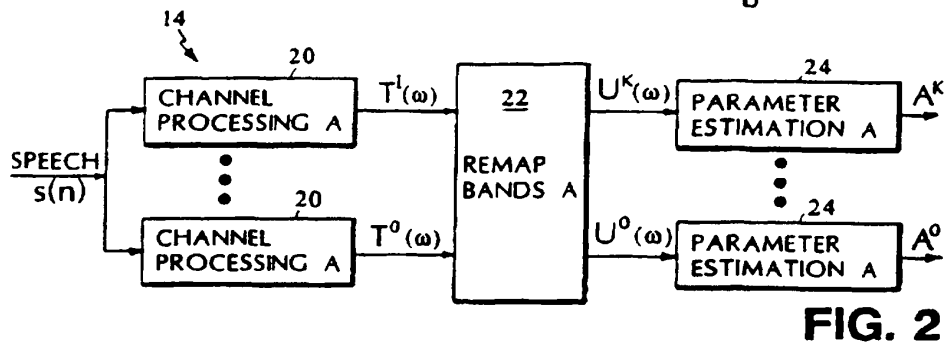
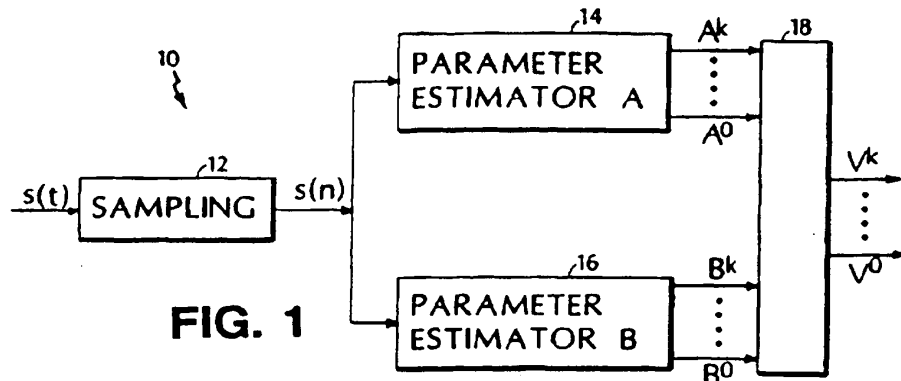
25. A system for analysing a digitized speech signal to determine excitation parameters for the digitized speech signal, comprising: means for dividing the digitized speech signal into one or more frequency band signals; means for determining a first preliminary excitation parameter using a first method that includes performing a nonlinear operation on at least one of the frequency band signals to produce at least one modified frequency band signal and determining the first preliminary excitation parameter using the at least one modified frequency band signal; means for determining a second preliminary excitation parameter using a second method that is different from the above said first method; and means for using the first and second preliminary excitation parameters to determine an excitation parameter for the digitized speech signal.

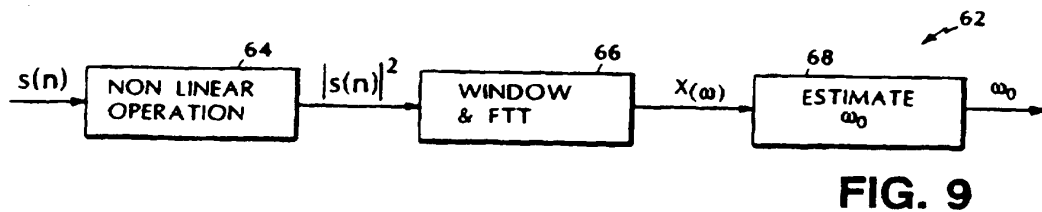
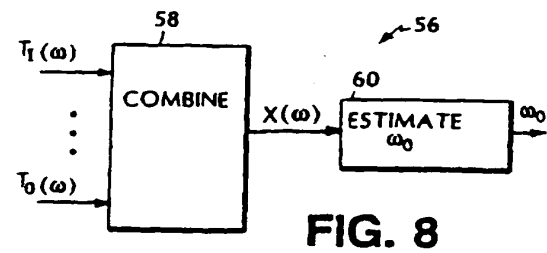
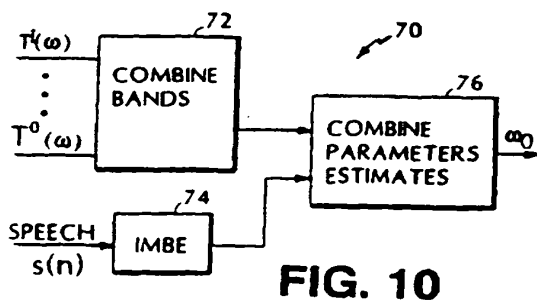
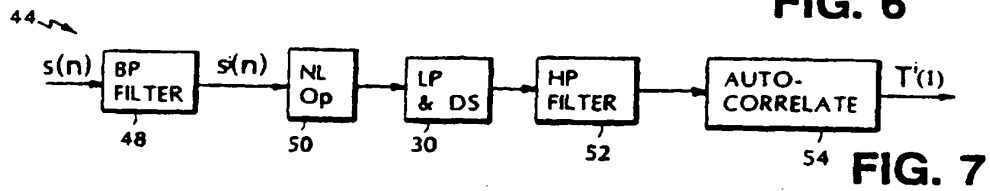
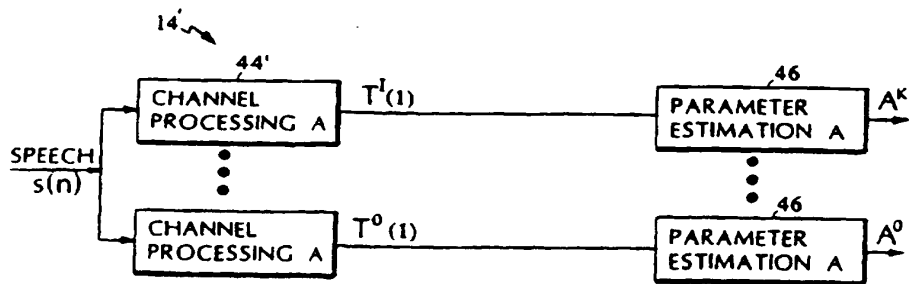
26. A system for analysing a digitized speech signal to determine excitation parameters for the digitized speech signal, comprising: means for determining preliminary excitation parameters from the digitized speech signal; and means for smoothing the preliminary excitation parameters to produce excitation parameters.

27. A system for analysing a digitized speech signal to determine modified excitation parameters for the digitized speech signal, comprising: means for estimating a fundamental frequency for the digitized speech signal; means for evaluating a voiced/unvoiced function using the estimated fundamental frequency to produce a first preliminary voiced/unvoiced parameter; means for evaluating the voiced/unvoiced function using another frequency derived from the estimated fundamental frequency to produce a second preliminary voiced/unvoiced parameter; and means for combining the first and second preliminary voiced/unvoiced parameters to produce a voiced/unvoiced parameter.

28. A system for analysing a digitized speech signal to determine a fundamental frequency estimate for the digitized speech signal, comprising: means for determining a predicted fundamental frequency estimate from previous fundamental frequency estimates; means for determining an initial fundamental frequency estimate; means for evaluating an error function at the initial fundamental frequency estimate to produce a first error function value; means for evaluating the error function at at least one other frequency derived from the initial fundamental frequency estimate to produce a second error function value; and means for selecting a fundamental frequency estimate using the predicted fundamental frequency estimate, the initial fundamental frequency estimate, the first error function value, and the second error function value.

29. A method of analysing a digitized speech signal to determine a voiced/unvoiced function for the digitized speech signal, comprising: dividing the digitized speech signal into at least two frequency band signals; determining a first preliminary voiced/unvoiced function for at least two of the frequency band signals using a first method; determining a second preliminary voiced/unvoiced function for at least two of the frequency band signals using a second method which is different from the above said first method; and using the first and second preliminary excitation parameters to determine a voiced/unvoiced function for at least two of the frequency band signals.





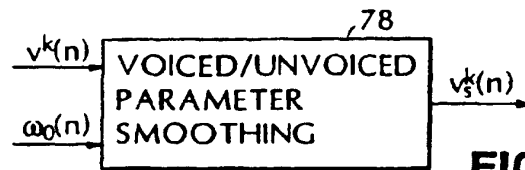


FIG. 11

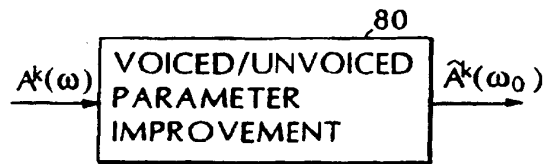


FIG. 12

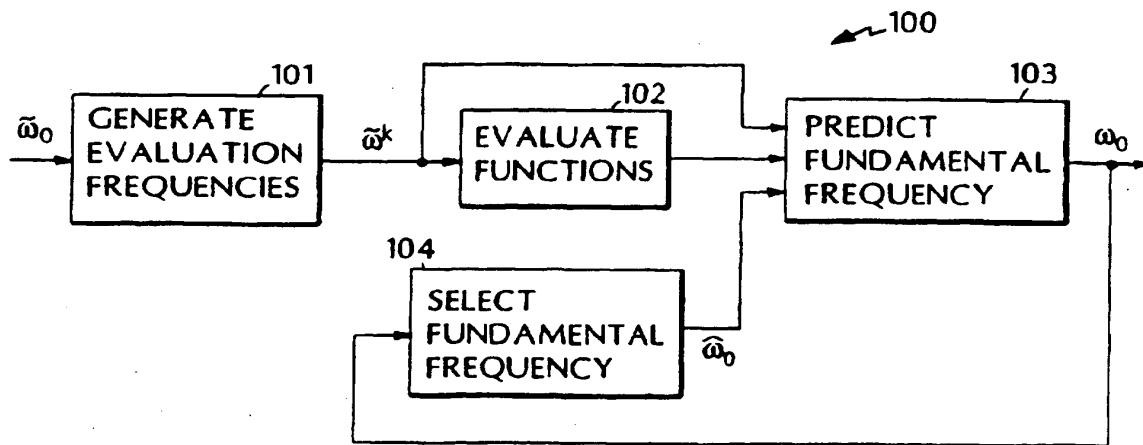


FIG. 13

(19)



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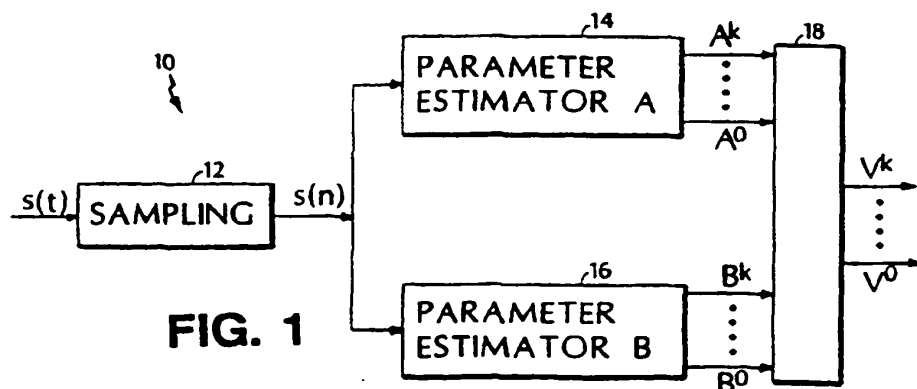
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(54) Estimation of excitation parameters

(57) Excitation parameters for a digitized speech signal are determined by analysing the digitized speech signal. The digitized speech signal is divided into at least two frequency bands. A first preliminary excitation parameter is determined by performing a nonlinear operation on at least one of the frequency band signals to produce a modified frequency band signal and determining the first preliminary excitation parameter using the modified frequency band signal. A second preliminary

excitation parameter is determined using a method different from the first method. The first and second preliminary excitation parameters are used to determine an excitation parameter for the digitized speech signal. The method is useful in encoding speech. Speech synthesized using the parameters estimated based on the invention generates high quality speech at various bit rates useful for applications such as satellite voice communication.



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European Patent
Office

EUROPEAN SEARCH REPORT

Application Number
EP 96 30 0245

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.6)
A	DELLER, PROAKIS, HANSEN: "Discrete-time processing of speech signals" 1993, MACMILLAN PUBLISHING COMPANY XP002051530 * page 460, paragraph 7.4.1 * * page 461; figure 7.25 *	1,17,25, 27,29	G10L9/14 G10L7/04
A	WO 88 07740 A (AMERICAN TELEPHONE & TELEGRAPH) * figure 1 * * abstract *	1,17,25, 27,29	
A	ICASSP 79. 1979 IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH AND SIGNAL PROCESSING, WASHINGTON, DC, USA, 2-4 APRIL 1979, 1979, NEW YORK, NY, USA, IEEE, USA, pages 69-72, XP002051529 KUREMATSU A ET AL: "A linear predictive vocoder with new pitch extraction and exciting source" * figure 1 * * page 70, column 1, line 3 - line 6 *	1,17,25, 27,29	
A	WO 92 05539 A (DIGITAL VOICE SYSTEMS INC) * page 9, line 1 - line 14 *	1,17,25, 27,29	
			TECHNICAL FIELDS SEARCHED (Int.Cl.6)
			G10L
The present search report has been drawn up for all claims			
Place of search THE HAGUE		Date of completion of the search 9 January 1998	Examiner Krembel, L
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 L : document cited for other reasons & : member of the same patent family, corresponding document			

EPO FORM 1503 03/82 (P04C01)



European Patent
Office

CLAIMS INCURRING FEES

EP 96300245.6

The present European patent application comprised at the time of filing more than ten claims.

- ☐ All claims fees have been paid within the prescribed time limit. The present European search report has been drawn up for all claims.
- ☐ Only part of the claims fees have been paid within the prescribed time limit. The present European search report has been drawn up for the first ten claims and for those claims for which claims fees have been paid, namely claims:
- ☐ No claims fees have been paid within the prescribed time limit. The present European search report has been drawn up for the first ten claims.

LACK OF UNITY OF INVENTION

The Search Division considers that the present European patent application does not comply with the requirement of unity of invention and relates to several inventions or groups of inventions.

namely:

See Sheet 3.

- ☐ All further search fees have been paid within the fixed time limit. The present European search report has been drawn up for all claims.
- ☐ Only part of the further search fees have been paid within the fixed time limit. The present European search report has been drawn up for those parts of the European patent application which relate to the inventions in respect of which search fees have been paid, namely claims:
- ☒ None of the further search fees has been paid within the fixed time limit. The present European search report has been drawn up for those parts of the European patent application which relate to the invention first mentioned in the claims.
- namely claims: 1-11, 17-20, 25, 27, 29



European Patent
Office

EUROPEAN SEARCH REPORT

Application Number
EP 96 30 0245

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.6)
A	IEEE TRANSACTIONS ON SIGNAL PROCESSING, vol. 39, no. 2, 1 February 1991, pages 319-329, XP000206434 KRUBSACK D A ET AL: "AN AUTOCORRELATION PITCH DETECTOR AND VOICING DECISION WITH CONFIDENCE MEASURES DEVELOPED FOR NOISE-CORRUPTED SPEECH" * figure 1 * * Paragraph II.B "Features for the Voicing Decision and Confidence Measures" * * Paragraph II.C "Voicing Decision" * * Paragraph II.D "Voicing Confidence" * -----	10	
			TECHNICAL FIELDS SEARCHED (Int.Cl.6)
The present search report has been drawn up for all claims			
Place of search THE HAGUE		Date of completion of the search 9 January 1998	Examiner Krembel, L
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 L : document cited for other reasons & : member of the same patent family, corresponding document			

EPO FORM 1503 03 82 (P04C01)



European Patent
Office

LACK OF UNITY OF INVENTION
SHEET B

Application Number
EP 96 30 0245

The Search Division considers that the present European patent application does not comply with the requirements of unity of invention and relates to several inventions or groups of inventions, namely:

1. Claims: 1-11,17-20,25,27,29

Determination of excitation parameters using a combination of two preliminary excitation parameters.

2. Claims: 12-16,26

Smoothing of excitation parameters.

3. Claims: 21-24,28

Fundamental frequency tracking method.

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0 421 444 A2

EUROPEAN PATENT APPLICATION

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Ⓢ! Int. Cl.⁵: **G10L 9/14**

② Date of filing: 05.10.90

③ Priority: 05.10.89 JP 260531/89

④⁴³ Date of publication of application:
10.04.91 Bulletin 91/15

⑧4 Designated Contracting States:
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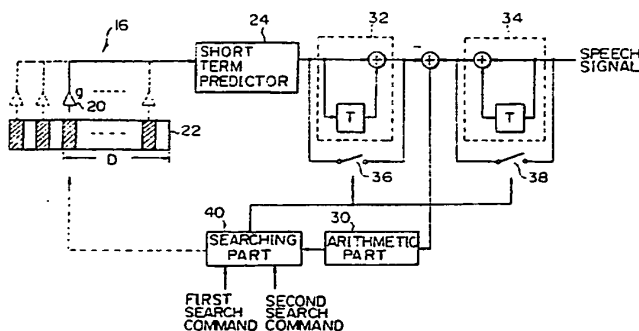
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⑤4 Pitch period searching method and circuit for speech code.

⑤7 Pitch periods for a long term predictor (16) included in a speech codec are searched in two searching stages. In the first searching stage, probable pitch periods are searched skipping a constant number of pitch periods, and in the second searching stage, pitch periods including the pitch period determined in the first searching stage and pitch periods neighboring the pitch period on both sides are searched.

Fig. 4



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EP 0 421 444 A2

PITCH PERIOD SEARCHING METHOD AND CIRCUIT FOR SPEECH CODEC

BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention relates to a method and a circuit for searching for a pitch period of a speech signal to determine coefficients for a long term predictor which is used in a coder and decoder (codec) for speech signals.

2. Description of the Related Art

Recently, high performance speech coding, wherein speech signals can be transmitted at low bit rates without remarkably degrading quality of the speech signals, have been required in local communication systems, digital mobile communication systems, and the like.

In several types of speech coding, for example, code-excited linear predictive coding (CELP), residual-excited linear predictive coding (RELP), and multi-pulse excited linear predictive coding (MPC), a long term predictor (pitch predictor) is used for performing long term prediction based on periodicity of a speech signal.

Coefficients for the long term predictor are determined by minimizing a total squared prediction error after pitch prediction. Accordingly, the total squared prediction error for all pitch periods which are probable in speech signals had to be estimated to find the most adequate coefficients for each speech signal block. Therefore, the number of arithmetic operations becomes enormous and the scale of required hardware becomes large.

SUMMARY OF THE INVENTION

It is an object of the present invention to provide a method and a circuit which require a relatively small number of arithmetic operations and relatively small size hardware.

In accordance with the present invention there is provided a pitch period searching method for searching pitch periods, which are probable in speech signals, for the most adequate pitch period for a long term predictor included in a speech codec, characterized in that the method comprises the steps of: first searching the probable pitch periods skipping a first number of pitch periods, to find the most adequate pitch period among the searched pitch periods, and second searching a second number of pitch periods including the pitch period and pitch periods neighboring the pitch period on both sides, to find the most adequate pitch period among the second number of pitch periods.

In accordance with the present invention there is also provided a pitch period searching circuit for searching pitch periods which are probable in speech signals for the most adequate pitch period for a long term predictor included in a speech codec, comprising arithmetic means for estimating suitability of the pitch period, characterized in that the circuit further comprises: searching means for searching the probable pitch periods skipping a first number of pitch periods, to find the most adequate pitch period among the searched pitch periods based on estimation by the arithmetic means, in response to a first search command, and for searching a second number of pitch periods including the above pitch period and pitch periods neighboring the pitch period on both sides, to find the most adequate pitch period among the second number of pitch periods based on estimation by the arithmetic means, in response to a second search command.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 is a block diagram representing a general construction of a CELP coder as an example of speech coders having a long term predictor;

Figure 2 is a block diagram representing a conventional searching process for pitch periods for the long term predictor;

3A is a block diagram representing a first searching stage according to the present invention;
 Figure 3B is a block diagram representing a second searching stage according to the present invention;
 and
 Figure 4 is a block diagram showing a more concrete and more detailed example of the present invention.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

Before describing the preferred embodiments according to the invention, examples of aforementioned related art are given with reference to the accompanying drawings.

Figure 1 is a block diagram showing a general construction of a speech coder using CELP as an example of speech coders having a long term predictor 16.

A plurality of stochastic signals are stored in a codebook 10. One of the stochastic signals is selected by a switch 12 according to a number i , is multiplied by a coefficient b in a multiplier 14, and passes through the long term predictor 16 and a short term predictor 24. A prediction error is estimated by subtracting the output of the short term predictor 24 from a speech signal in a subtractor 26. Coefficients for the short term predictor 24 are determined by LPC analysis of the speech signal. Also, the number i , gain b , gain g of a multiplier 20 in the long term predictor 16, and delay time D of a shift register 22 in the long term predictor 16 are determined by minimizing the total squared prediction error over a speech signal block. These coefficients are transmitted as a code block representing the speech signal block.

In a decoder side, the speech signal blocks are sequentially reproduced based on received code blocks, and thus speech signals are reproduced.

Figure 2 shows a block diagram representing a conventional method for determining values of g and D for the long term predictor 16.

Usually, in pitch period search, output of the codebook 10 is set to zero to avoid joint optimization of pitch and codebook parameter which require enormous computation.

Past excitation signals v for the short term predictor are stored in the shift register 22. The excitation signals v_{i-D} ($i = 1, 2 \dots N$, where N is length of a signal block), which are D delayed signals, are taken out from the shift register 22, multiplied by gain g in the multiplier 20, and input to the short term predictor 24. The relationship between output $g \cdot y_i$ and input $g \cdot v_{i-D}$ of the short term predictor 24 is expressed by following equation:

$$g \cdot y_i = g \cdot v_{i-D} + g \cdot \sum_{j=1}^p a_j y_{i-j}$$

or

$$y_i = v_{i-D} + \sum_{j=1}^p a_j y_{i-j} \quad \dots (1),$$

wherein a_j ($j = 1, 2 \dots p$) are linear prediction coefficients for the short term predictor 24 and p is the order of the short term predictor 24.

The total squared prediction error E_D over a speech signal block is calculated from the following equation:

$$E_D = \sum_{i=1}^N (x_i - g \cdot y_i)^2 \quad \dots (2),$$

wherein x_i is a sample value of the speech signal.

A gain g which minimizes the E_D is obtained from the following equation:

$$\partial E_D / \partial g = 0$$

$$2 \sum_{i=1}^N (x_i - g \cdot y_i) y_i = 0$$

Therefore,

$$g = \frac{\sum_{i=1}^N x_i \cdot y_i}{\sum_{i=1}^N y_i^2} \quad \dots (3)$$

Substituting equation (3) into equation (2),

$$E_D = \sum_{i=1}^N x_i^2 - \left(\sum_{i=1}^N x_i \cdot y_i \right)^2 / \sum_{i=1}^N y_i^2 \quad \dots (4)$$

is obtained. Replacing the second term of the equation (4) by A, namely,

$$A = \left(\sum_{i=1}^N x_i \cdot y_i \right)^2 / \sum_{i=1}^N y_i^2 \quad \dots (5)$$

the total squared prediction error E_D is minimized when A is maximum.

A searching part 28 sequentially selects one of all probable pitch periods for the delay time D, and an arithmetic part 30 estimates the total squared prediction error E_D for each delay time D.

As mentioned above, in the conventional pitch period searching method, enormous operation according to the equation (5) for all probable pitch periods is required, and therefore, a scale of required hardware becomes large.

The preferred embodiments of the present invention will now be described with reference to the accompanying drawings.

The pitch period searching process according to the present invention includes a first searching stage and a second searching stage. Figure 3A shows the first searching stage.

The first searching process is performed skipping M samples wherein M is a constant value, and then a pitch period generating the least total square prediction error is determined. Therefore, the number of arithmetic operations is remarkably decreased. But, as skipped samples are increased, correlation between neighboring samples becomes weak. To avoid this, smoothing parts 32 and 34 are provided as shown by dashed lines. Both of the smoothing parts 32 and 34 have a smoothing factor M, and smooth output signals of the short term predictor 24 and the speech signals, respectively, so that the searching accuracy is improved.

Figure 3B shows the second searching stage. In the second searching stage, a predetermined number of samples neighboring the pitch period determined in the first searching stage on both sides are searched for a pitch period generating the least total squared prediction error, so that the most adequate pitch period is finally determined.

Figure 4 shows a more concrete and more detailed example of the present invention, but the present invention is not restricted to the example. In this example, pitch periods are searched within a range of 20 to 147 sampling intervals. The first searching process is performed skipping one sample. The smoothing parts 32 and 34 calculate moving averages of two neighboring samples of the output of the short term predictor 24 and the speech signals, respectively. Switches 36 and 38 which are controlled by the searching part 40 are provided in order to bypass the smoothing parts 32 and 34 in the second searching process.

When the searching part 40 receives a first search command, the searching part 40 opens the switches 36 and 38, sequentially sets a taking-out position of the shift register 22 at 20, 22, 24 ... samples delay

positions. The arithmetic part 30 calculates the total squared prediction error for each position, and a pitch period D_1 which generates the least total squared prediction error is determined in the searching part 40.

Next, when the searching part 40 receives a second search command, the searching part 40 closes the switches 36 and 36 by bypass the smoothing parts 32 and 34, and then searches the pitch period D_1 and each of two pitch periods neighboring the pitch period D_1 on both sides to find a pitch period D_2 which generates the least total squared prediction error among the five searched pitch periods. The pitch period D_2 is finally determined as the most adequate pitch period.

Reference signs in the claims are intended for better understanding and shall not limit the scope.

10

Claims

1. A pitch period searching method for searching pitch periods, which are probable in speech signals, for the most adequate pitch period for a long term predictor (16) included in a speech codec, characterized in that the method comprises the steps of:

15 first searching the probable pitch periods skipping a first number (M) of pitch periods, to find the most adequate pitch period (D_1) among the searched pitch periods, and
second searching a second number of pitch periods including said pitch period (D_1) and pitch periods neighboring said pitch period (D_1) on both sides, to find the most adequate pitch period (D_2) among the
20 second number of pitch periods.

2. A searching method as claimed in claim 1, wherein said speech codec further includes a short term predictor (24), and said first and second searching steps comprise the steps of:

estimating a total squared prediction error between a speech signal and a predictive signal thereof predicted with said long term predictor (16) and said short term predictor (24), for each searched pitch
25 period, and

selecting a pitch period which generates the least total squared prediction error among the searched pitch periods for the most adequate pitch period (D_1 , D_2).

3. A searching method as claimed in claim 2, wherein said estimating step included in said first searching step comprises a step of smoothing said speech signal and said predictive signal at a time constant
30 corresponding to said first number (M), before calculating said prediction error.

4. A pitch period searching circuit for searching pitch periods which are probable in speech signals for the most adequate pitch period for a long term predictor (16) included in a speech codec, comprising arithmetic means (30) for estimating suitability of the pitch period, characterized in that the circuit further comprises:

35 searching means (40) for searching the probable pitch periods skipping a first number (M) of pitch periods, to find the most adequate pitch period (D_1) among the searched pitch periods based on estimation by said arithmetic means (30), in response to a first search command, and for searching a second number of pitch periods including said pitch period (D_1) and pitch periods neighboring said pitch period (D_1) on both sides, to find the most adequate pitch period (D_2) among the second number of pitch periods based on estimation by said arithmetic means (30), in response to a second search command.

40 5. A searching circuit as claimed in claim 4, wherein said speech codec further comprises a short term predictor (24), said arithmetic means (30) estimates a total squared prediction error between a speech signal and a predictive signal thereof predicted with said long term predictor (16) and said short term predictor (24), and said searching means (40) select the most adequate pitch period (D_1 , D_2) which generates the least total squared prediction error among the searched pitch periods.

45 6. A searching circuit as claimed in claim 5, further comprises
two smoothing means (32, 34) for smoothing said speech signal and said predictive signal, respectively, at a time constant corresponding to said first number (M),

and two switch means (36, 38) for bypassing said two smoothing means (32, 34), respectively, wherein said searching means (40) open said switch means (36, 38) in response to said first search command, and
50 close said switch means (36, 38) in response to said second search command.

55

Fig. 1

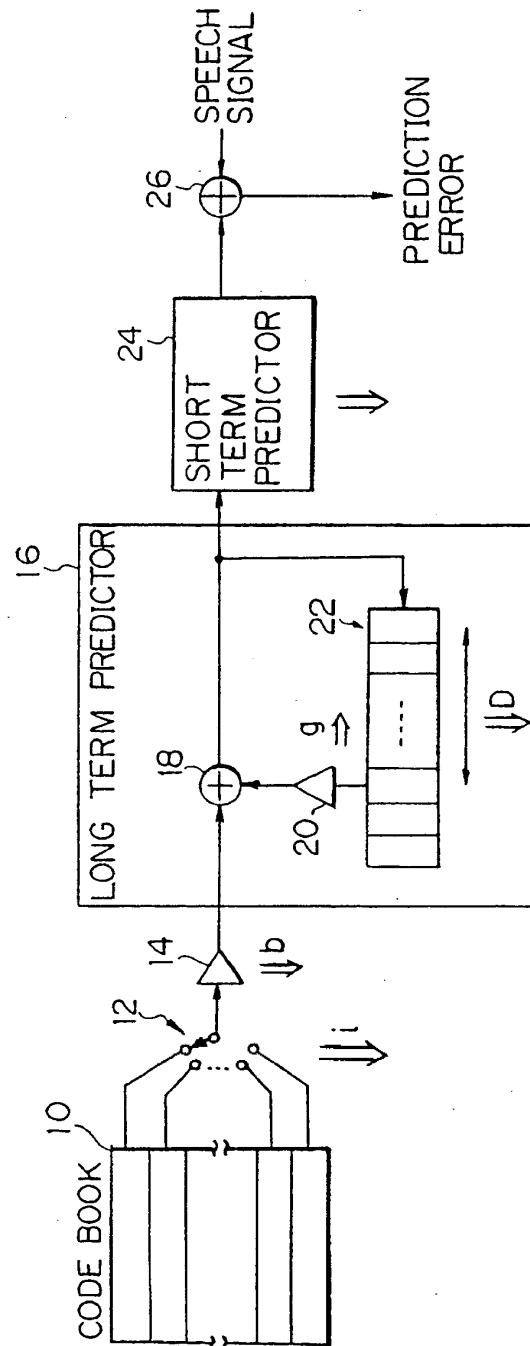


Fig. 2

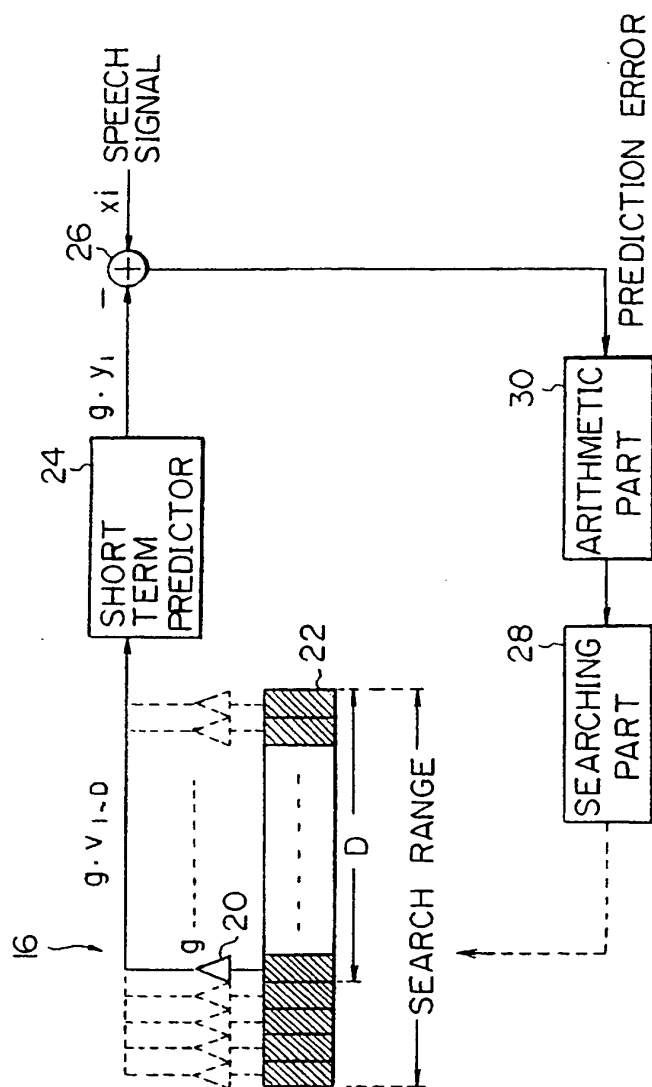


Fig. 3A

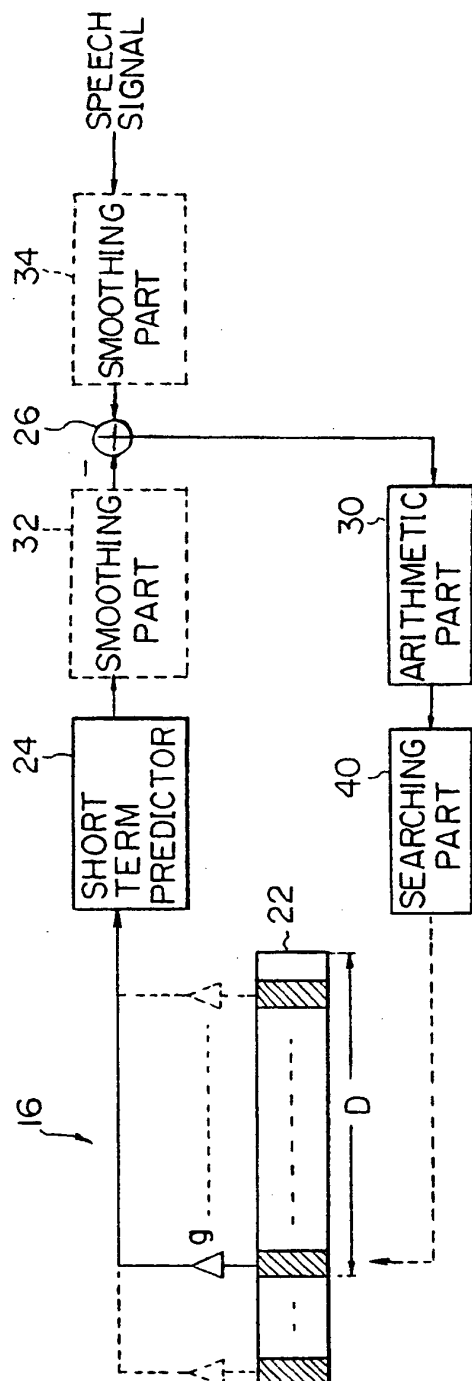


Fig. 3B

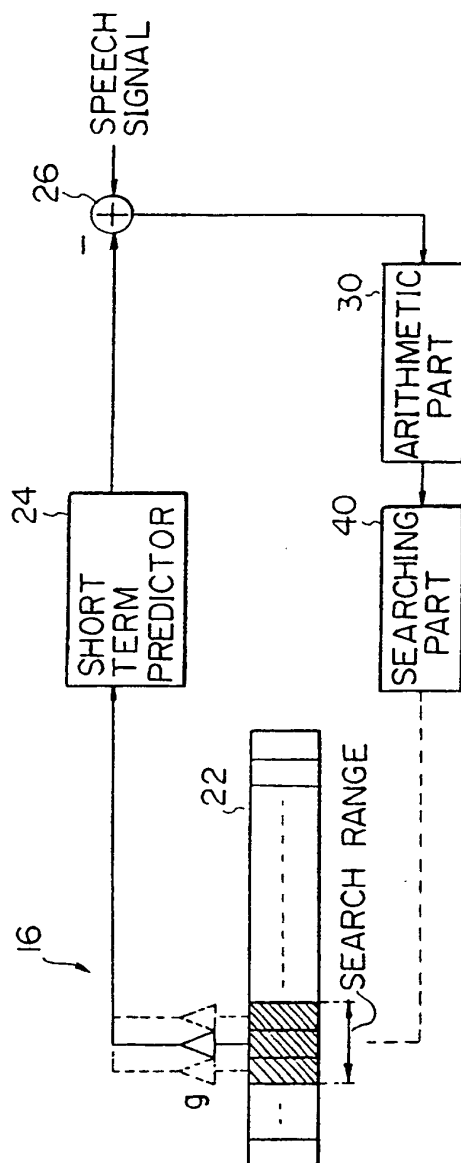
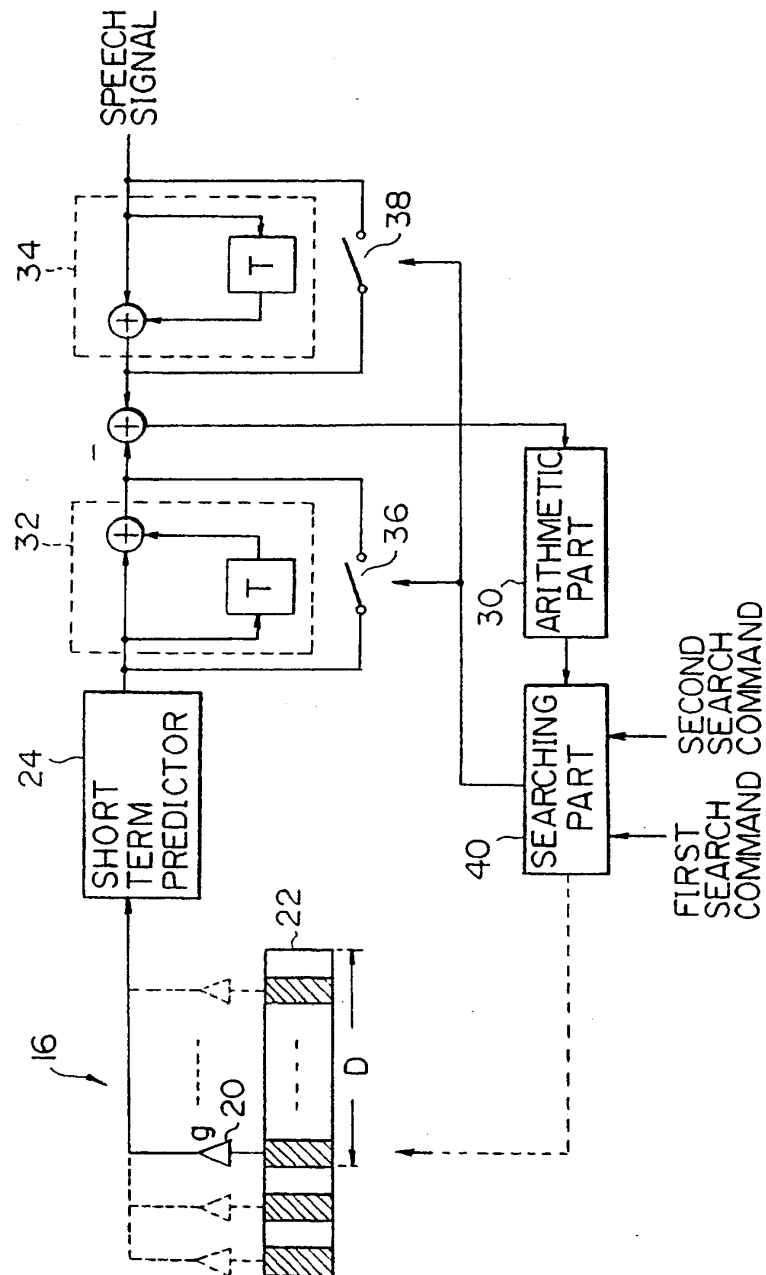


Fig. 4





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⑫ **EUROPEAN PATENT APPLICATION**

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⑤① Int. Cl.⁵: **G10L 9/14**

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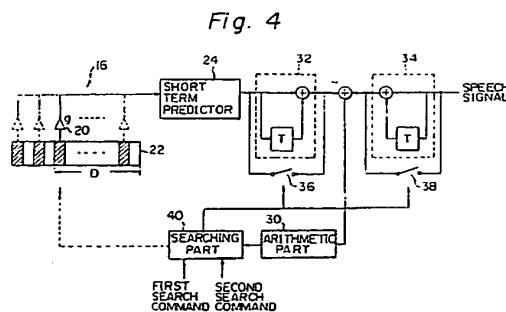
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W-8000 München 81(DE)

⑤④ Pitch period searching method and circuit for speech code.

⑤⑦ Pitch periods for a long term predictor (16) included in a speech codec are searched in two searching stages. In the first searching stage, probable pitch periods are searched skipping a constant number of pitch periods, and in the second searching stage, pitch periods including the pitch period determined in the first searching stage and pitch periods neighboring the pitch period on both sides are searched.



EP 0 421 444 A3



European
Patent Office

EUROPEAN SEARCH REPORT

Application Number

EP 90 11 9097

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)
X	EP-A-0 331 857 (INTERNATIONAL BUSINESS CORP.) * Page 2, line 24 - page 3, line 31 * -----	1,2,4,5	G 10 L 9/14
			TECHNICAL FIELDS SEARCHED (Int. Cl.5)
			G 10 L 9/14
The present search report has been drawn up for all claims			
Place of search The Hague		Date of completion of search 10 May 91	Examiner ARMSPACH J.F.A.M.
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 L : document cited for other reasons ----- & : member of the same patent family, corresponding document			



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U.S. APPLICATION NO.

FIRST NAMED APPLICANT

ATTY. DOCKET NO.

09/830114

BESSETTE

B

4082-0129P

INTERNATIONAL APPLICATION NO.

PCT/CA98/01008

BIRCH STEWART KOLASCH & BIRCH
PO BOX 747
FALLS CHURCH, VA 22040 0747

I.A. FILING DATE

PRIORITY DATE

27 OCT 99

27 OCT 98

DATE MAILED:

22 MAY 2001

NOTIFICATION OF MISSING REQUIREMENTS UNDER 35 U.S.C. 371 IN THE UNITED STATES DESIGNATED/ELECTED OFFICE (DO/EO/US)

1. The following items have been submitted by the applicant or the IB to the United States Patent and Trademark Office as ☐ a Designated Office (37 CFR 1.494) ☒ an Elected Office (37 CFR 1.495):

- ☒ U.S. Basic National Fee. ☒ Indication of Small Entity Status.
☒ Copy of the international application. ☐ Translation of the international application into English.
☐ Oath or Declaration of inventor(s). ☐ Translation of Article 19 amendments into English.
☐ Copy of Article 19 amendments. ☐ Other:
☒ Priority Document.
☒ The International Preliminary Examination Report in English and its Annexes, if any.
☐ Translation of Annexes to the International Preliminary Examination Report into English.

2. ☒ Applicant has requested early processing under 35 U.S.C. 371(f) but has not filed the following indicated items and/or the indicated items in paragraph 3 below. The Basic National Fee and the copy of the international application must be filed prior to 20 or 30 months from the priority date to avoid abandonment.

- ☐ U.S. Basic National Fee. ☐ Copy of the international application.

3. The following items **MUST** be furnished within the period set forth below in order to complete the requirements for acceptance under 35 U.S.C. 371:

- ☐ a. Translation of the application into English. A processing fee will be required if submitted later than the appropriate 20 or 30 months from the priority date.
- The current translation is defective for the reasons indicated on the attached Notice of Defective Translation.
☐ b. Processing fee for providing the translation of the application and/or the Annexes later than the appropriate 20 or 30 months from the priority date (37 CFR 1.492(f)).
☒ c. Oath or declaration of the inventors, in compliance with 37 CFR 1.497(a) and (b), properly identifying the application (preferably by the International application number and international filing date). A surcharge will be required if submitted later than the appropriate 20 or 30 months from the priority date.
- The current oath or declaration does not comply with 37 CFR 1.497(a) and (b) for the reasons indicated on the attached PCT/DO/BO/917.
☐ d. Surcharge for providing the oath or declaration later than the appropriate 20 or 30 months from the priority date (37 CFR 1.492(e)).

4. Additional claim fees of \$_____ as a ☐ large entity ☐ small entity, including any required multiple dependent claim fee, are required. Applicant must submit the additional claim fees or cancel the additional claims for which fees are due (37 CFR 1.492(g)). See attached PTO-875.

5. ☐ Applicant has not submitted the required sequence listing pursuant to 37 CFR 1.821-1.825. See attached PCT/DO/BO/920.

ALL OF THE ITEMS SET FORTH IN 3(a)-3(d), 4 AND 5 ABOVE MUST BE SUBMITTED WITHIN TWO (2) MONTHS FROM THE DATE OF THIS NOTICE OR BY 22 OR 32 MONTHS (where 37 CFR 1.495 applies) FROM THE PRIORITY DATE FOR THE APPLICATION, WHICHEVER IS LATER. FAILURE TO PROPERLY RESPOND WILL RESULT IN ABANDONMENT.

The time period set above may be extended by filing a petition and fee for extension of time under the provisions of 37 CFR 1.136(a).

6. If box 3a or 3c is checked, a translation of the Annexes **MUST** be submitted no later than the time period set above or the Annexes will be cancelled. A processing fee will be required if submitted later than 20 or 30 months from the priority date.
7. ☐ The Article 19 amendments are cancelled since a translation was not provided by the appropriate 20 (37 CFR 1.494(d)) or 30 (37 CFR 1.495(d)) months from the priority date.

Applicant is reminded that any communication to the United States Patent and Trademark Office must be mailed to the address given in the heading and include the U.S. application no. shown above. (37 CFR 1.5)

A copy of this notice *MUST* be returned with this response.

Enclosed: ☐ PCT/DO/BO/917
☐ PTO-875

☐ Notice of Defective Translation
☐ PCT/UX/BO/920

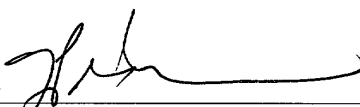
Anita D. Johnson

FORM PCT/DO/BO/905 (March 2001)

Telephone: 703-305-3661

J007 Rec'd PCT/PTO 2 4 APR 2001

FORM PTO-1390 (REV. 11-2000)		U.S. DEPARTMENT OF COMMERCE PATENT AND TRADEMARK OFFICE	ATTORNEY'S DOCKET NUMBER 4082-0129P
TRANSMITTAL LETTER TO THE UNITED STATES DESIGNATED/ELECTED OFFICE (DO/EO/US) CONCERNING A FILING UNDER 35 U.S.C. 371			U.S. APPLICATION NO. (If known, see 37 CFR 1.5) 09/830114 NEW
INTERNATIONAL APPLICATION NO. PCT/CA99/01008	INTERNATIONAL FILING DATE October 27, 1999	PRIORITY DATE CLAIMED October 27, 1998	
TITLE OF INVENTION A METHOD AND DEVICE FOR ADAPTIVE BANDWIDTH PITCH SEARCH IN CODING WIDEBAND SIGNALS			
APPLICANT(S) FOR DO/EO/US Bruno BESSETTE, Redwan SALAMI, Roch LEFEBVRE			
Applicant herewith submits to the United States Designated/Elected Office (DO/EO/US) the following items and other information:			
1. ☒ This is a FIRST submission of items concerning a filing under 35 U.S.C. 371. 2. ☐ This is a SECOND or SUBSEQUENT submission of items concerning a filing under 35 U.S.C. 371. 3. ☒ This express request to begin national examination procedures (35 U.S.C. 371(f)) at any time rather than delay examination until the expiration of the applicable time limit set in 35 U.S.C. 371(b) and PCT Articles 22 and 39 (1). 4. ☐ The US has been elected by the expiration of 19 months from the priority date (Article 31). 5. ☒ A copy of the International Application as filed (35 U.S.C. 371(c)(2)) a. ☒ is transmitted herewith (required only if not transmitted by the International Bureau). b. ☒ has been transmitted by the International Bureau. c. ☐ is not required, as the application was filed in the United States Receiving Office (RO/US). 6. ☐ An English language translation of the International Application as filed (35 U.S.C. 371(c)(2)). a. ☐ is transmitted herewith. b. ☐ has been previously submitted under 35 U.S.C. 154(d)(4) 7. ☐ Amendments to the claims of the International Application under PCT Article 19 (35 U.S.C. 371(c)(3)). a. ☐ are transmitted herewith (required only if not transmitted by the International Bureau). b. ☐ have been transmitted by the International Bureau. c. ☐ have not been made; however, the time limit for making such amendments has NOT expired. d. ☐ have not been made and will not be made. 8. ☐ An English language translation of the amendments to the claims under PCT Article 19 (35 U.S.C. 371(c)(3)). 9. ☐ An oath or declaration of the inventor(s) (35 U.S.C. 371(c)(4)). 10. ☐ An English language translation of the annexes of the International Preliminary Examination Report under PCT Article 36 (35 U.S.C. 371(c)(5)).			
Items 11. to 20. below concern document(s) or information included:			
11. ☒ An Information Disclosure Statement under 37 CFR 1.97 and 1.98./International Search Report with cited references 12. ☐ An assignment document for recording. A separate cover sheet in compliance with 37 CFR 3.28 and 3.31 is included. 13. ☒ A FIRST preliminary amendment. 14. ☐ A SECOND or SUBSEQUENT preliminary amendment. 15. ☐ A substitute specification. 16. ☐ A change of power of attorney and/or address letter. 17. ☐ A computer-readable form of the sequence listing in accordance with PCT Rule 13ter.2 and 35 U.S.C. 1.821-1.825. 18. ☐ A second copy of the published international application under 35 U.S.C. 154(d)(4). 19. ☐ A second copy of the English language translation of the international application under 35 U.S.C. 154(d)(4). 20. ☒ Other items or information:			

U.S. APPLICATION NO (if known, see 37 CFR 1.5) 09/830114		INTERNATIONAL APPLICATION NO PCT/CA99/01008		ATTORNEY'S DOCKET NUMBER 4082-0129P	
21. ☒ The following fees are submitted:				CALCULATIONS PTO USE ONLY	
BASIC NATIONAL FEE (37 CFR 1.492(a)(1)-(5): Neither international preliminary examination fee (37 CFR 1.482) nor international search fee (37 CFR 1.445(a)(2)) paid to USPTO and International Search Report not prepared by the EPO or JPO.				\$1,000.00	
International preliminary examination fee (37 CFR 1.482) not paid to USPTO but International Search Report prepared by the EPO or JPO				\$860.00	
International preliminary examination fee (37 CFR 1.482) not paid to USPTO but international search fee (37 CFR 1.445(a)(2)) paid to USPTO.				\$710.00	
International preliminary examination fee (37 CFR 1.482) paid to USPTO but all claims did not satisfy provisions of PCT Article 33(1)-(4)				\$690.00	
International preliminary examination fee (37 CFR 1.482) paid to USPTO and all claims satisfied provisions of PCT Article 33(1)-(4)				\$100.00	
ENTER APPROPRIATE BASIC FEE AMOUNT =				\$ 860.00	
Surcharge of \$130.00 for furnishing the oath or declaration later than ☐ 20 ☒ 30 months from the earliest claimed priority date (37 CFR 1.492(e)).				\$ 130.00	
CLAIMS	NUMBER FILED	NUMBER EXTRA	RATE		
Total Claims	63 - 20 =	43	X \$18.00	\$ 774.00	
Independent Claims	2 - 3 =	0	X \$80.00	\$ 0	
MULTIPLE DEPENDENT CLAIM(S) (if applicable) 0			+ \$270.00	\$ 0	
TOTAL OF ABOVE CALCULATIONS =				\$ 1764.00	
☒ Applicant claims small entity status. See 37 CFR 1.27. The fees indicated above are reduced by 1/2.				\$ (882.00)	
SUBTOTAL =				\$ 882.00	
Processing fee of \$130.00 for furnishing the English translation later than ☐ 20 ☐ 30 months from the earliest claimed priority date (37 CFR 1.492(f)).				\$ 0	
TOTAL NATIONAL FEE =				\$ 0	
Fee for recording the enclosed assignment (37 CFR 1.21(h)). The assignment must be accompanied by an appropriate cover sheet (37 CFR 3.28, 3.31). \$40.00 per property +				\$ 0	
TOTAL FEES ENCLOSED =				\$ 882.00	
				Amount to be: refunded	\$
				charged	\$
a. ☒ A check in the amount of \$ 882.00 to cover the above fees is enclosed. b. ☐ Please charge my Deposit Account. No. _____ in the amount of \$ _____ to cover the above fees. A duplicate copy of this sheet is enclosed. c. ☒ The Commissioner is hereby authorized to charge any additional fees which may be required, or credit any overpayment to Deposit Account No. <u>02-2448</u> . NOTE: Where an appropriate time limit under 37 CFR 1.494 or 1.495 has not been met, a petition to revive (37 CFR 1.137(a) or (b)) must be filed and granted to restore the application to pending status. Send all correspondence to: Birch, Stewart, Kolasch & Birch, LLP or Customer No. 2292 P.O. Box 747 Falls Church, VA 22040-0747 (703)205-8000 Date: <u>April 24, 2001</u> By  F. Prince Butler, #25,666					

09/830114
JC18 Rec'd PCT/PTC 2 4 APR 2001

PATENT
4082-0129P

IN THE U.S. PATENT AND TRADEMARK OFFICE

Applicant: Bruno BESSETTE et al. Conf.:
Int'l. Appl. No.: PCT/CA99/01008
Appl. No.: NEW Group:
Filed: April 24, 2001 Examiner:
For: A METHOD AND DEVICE FOR ADAPTIVE
BANDWIDTH PITCH SEARCH IN CODING
WIDEBAND SIGNALS

PRELIMINARY AMENDMENT

BOX PATENT APPLICATION

Assistant Commissioner for Patents
Washington, DC 20231

April 24, 2001

Sir:

The following Preliminary Amendments and Remarks are respectfully submitted in connection with the above-identified application.

AMENDMENTS

IN THE SPECIFICATION:

Please amend the specification as follows:

Page 1, after the heading "BACKGROUND OF THE INVENTION", insert the paragraph --This application is the national phase under 35 U.S.C. § 371 of PCT International Application No. PCT/CA99/01008 which has an International filing date of October 27, 1999, which designated the United States of America and was published in English.

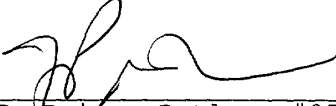
REMARKS

The specification has been amended to provide a cross-reference to the previously filed International Application.

If necessary, the Commissioner is hereby authorized in this, concurrent, and future replies, to charge payment or credit any overpayment to Deposit Account No. 02-2448 for any additional fees required under 37 C.F.R. § 1.16 or under 37 C.F.R. § 1.17; particularly, extension of time fees.

Respectfully submitted,

BIRCH, STEWART, KOLASCH & BIRCH, LLP

By 
F. Prince Butler, #25,666

FPB/tm
4082-0129P

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(Rev. 02/12/01)

4/PRTS

09/830114
JC18 Rec'd PCT/PTO 2 4 APR 2001

1

A METHOD AND DEVICE FOR ADAPTIVE BANDWIDTH

PITCH SEARCH IN CODING WIDEBAND SIGNALS

5

BACKGROUND OF THE INVENTION

1. Field of the invention:

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The present invention relates to an efficient technique for digitally encoding a wideband signal, in particular but not exclusively a speech signal, in view of transmitting, or storing, and synthesizing this wideband sound signal. More specifically, this invention deals with an improved pitch search device and method.

15

2. Brief description of the prior art:

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The demand for efficient digital wideband speech/audio encoding techniques with a good subjective quality/bit rate trade-off is increasing for numerous applications such as audio/video teleconferencing, multimedia, and wireless applications, as well as Internet and packet network applications. Until recently, telephone

25

bandwidths filtered in the range 200-3400 Hz were mainly used in speech coding applications. However, there is an increasing demand for wideband speech applications in order to increase the intelligibility and

naturalness of the speech signals. A bandwidth in the range 50-7000 Hz was found sufficient for delivering a face-to-face speech quality. For audio signals, this range gives an acceptable audio quality, but still lower than the CD quality which operates on the range 20-20000 Hz.

5 A speech encoder converts a speech signal into a digital bitstream which is transmitted over a communication channel (or stored in a storage medium). The speech signal is digitized (sampled and quantized with usually 16-bits per sample) and the speech encoder has the role of representing these digital samples with a smaller number of
10 bits while maintaining a good subjective speech quality. The speech decoder or synthesizer operates on the transmitted or stored bit stream and converts it back to a sound signal.

 One of the best prior art techniques capable of achieving a
15 good quality/bit rate trade-off is the so-called Code Excited Linear Prediction (CELP) technique. According to this technique, the sampled speech signal is processed in successive blocks of L samples usually called *frames* where L is some predetermined number (corresponding to 10-30 ms of speech). In CELP, a linear prediction (LP) filter is computed
20 and transmitted every frame. The L -sample frame is then divided into smaller blocks called *subframes* of size N samples, where $L=kN$ and k is the number of subframes in a frame (N usually corresponds to 4-10 ms of speech). An excitation signal is determined in each subframe, which usually consists of two components: one from the past excitation (also
25 called pitch contribution or adaptive codebook) and the other from an innovation codebook (also called fixed codebook). This excitation signal

is transmitted and used at the decoder as the input of the LP synthesis filter in order to obtain the synthesized speech.

5 An innovation codebook in the CELP context, is an indexed set of N -sample-long sequences which will be referred to as N -dimensional codevectors. Each codebook sequence is indexed by an integer k ranging from 1 to M where M represents the size of the codebook often expressed as a number of bits b , where $M=2^b$.

10 To synthesize speech according to the CELP technique, each block of N samples is synthesized by filtering an appropriate codevector from a codebook through time varying filters modeling the spectral characteristics of the speech signal. At the encoder end, the synthetic output is computed for all, or a subset, of the codevectors from the codebook (codebook search). The retained codevector is the one producing the synthetic output closest to
15 the original speech signal according to a perceptually weighted distortion measure. This perceptual weighting is performed using a so-called perceptual weighting filter, which is usually derived from the LP filter.

20 The CELP model has been very successful in encoding telephone band sound signals, and several CELP-based standards exist in a wide range of applications, especially in digital cellular applications. In the telephone band, the sound signal is band-limited to 200-3400 Hz and sampled at 8000 samples/sec. In wideband speech/audio applications, the sound signal is band-limited to 50-7000 Hz and sampled at 16000
25 samples/sec.

Some difficulties arise when applying the telephone-band optimized CELP model to wideband signals, and additional features need to be added to the model in order to obtain high quality wideband signals. Wideband signals exhibit a much wider dynamic range compared to telephone-band signals, which results in precision problems when a fixed-point implementation of the algorithm is required (which is essential in wireless applications). Further, the CELP model will often spend most of its encoding bits on the low-frequency region, which usually has higher energy contents, resulting in a low-pass output signal. To overcome this problem, the perceptual weighting filter has to be modified in order to suit wideband signals, and pre-emphasis techniques which boost the high frequency regions become important to reduce the dynamic range, yielding a simpler fixed-point implementation, and to ensure a better encoding of the higher frequency contents of the signal. Further, the pitch contents in the spectrum of voiced segments in wideband signals do not extend over the whole spectrum range, and the amount of voicing shows more variation compared to narrow-band signals. Therefore, in case of wideband signals, existing pitch search structures are not very efficient. Thus, it is important to improve the closed-loop pitch analysis to better accommodate the variations in the voicing level.

OBJECTS OF THE INVENTION

An object of the present invention is therefore to provide a method and device for efficiently encoding wideband (7000 Hz) sound signals using

CELP-type encoding techniques, using improved pitch analysis in order to obtain high a quality reconstructed sound signal.

SUMMARY OF THE INVENTION

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More specifically, in accordance with the present invention, there is provided a method for selecting an optimal set of pitch codebook parameters associated to a signal path, from at least two signal paths, having the lowest calculated pitch prediction error. The pitch prediction error is calculated in response to a pitch codevector from a pitch codebook search device. In at least one of the two signal paths, the pitch prediction error is filtered before supplying the pitch codevector for calculation of said pitch prediction error of said one path. Finally, the pitch prediction errors calculated in said at least two signal paths are compared, the signal path having the lowest calculated pitch prediction error is chosen, and the set of pitch codebook parameters associated to the choosen signal path are selected.

The pitch analysis device of the invention, for producing an optimal set of pitch codebook parameters, comprises:

a) at least two signal paths associated to respective sets of pitch codebook parameters, wherein:

- i) each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device; and
- ii) at least one of the two paths comprises a filter for filtering the pitch codevector before supplying the pitch

codevector to the path's pitch prediction error calculating device; and

- b) a selector for comparing the pitch prediction errors calculated in the signal paths, for choosing the signal path having the lowest calculated pitch prediction error, and for selecting the set of pitch codebook parameters associated to the chosen signal path.

The new method and device which achieve efficient modeling of the harmonic structure of the speech spectrum uses several forms of low pass filters applied to the past excitation and the one yielding higher prediction gain is selected. When subsample pitch resolution is used, the low pass filters can be incorporated into the interpolation filters used to obtain the higher pitch resolution.

In a preferred embodiment of the invention, each pitch prediction error calculating device of the pitch analysis device described above comprises:

a) a convolution unit for convolving the pitch codevector with a weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector;

and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

In another preferred embodiment of the invention, the pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^T y^{(j)} / \|y^{(j)}\|^2 \quad \text{where } j = 0, 1, 2, \dots, K, \text{ and } K \text{ corresponds to a number of signal paths,}$$

and where x is said pitch search target vector, and $y^{(j)}$ is said convolved pitch codevector.

The present invention further relates to an encoder, having the pitch analysis device described above, for encoding a wideband input signal and comprising:

- a) a linear prediction synthesis filter calculator responsive to the wideband signal for producing linear prediction synthesis filter coefficients;
- b) a perceptual weighting filter, responsive to the wideband signal and the linear prediction synthesis filter coefficients, for producing a perceptually weighted signal;
- c) an impulse response generator responsive to the linear prediction synthesis filter coefficients for producing a weighted synthesis filter impulse response signal;
- d) a pitch search unit for producing pitch codebook parameters, comprising:

i) a pitch codebook search device responsive to the perceptually weighted signal and the linear prediction synthesis filter coefficients for producing the pitch codevector and an innovative search target vector; and

ii) the pitch analysis device responsive to the pitch codevector for selecting, from the sets of pitch codebook parameters, the set of pitch codebook parameters associated to the path having the lowest calculated pitch prediction error;

d) an innovative codebook search device, responsive to the weighted synthesis filter impulse response signal, and the innovative search target vector, for producing innovative codebook parameters; and

e) a signal forming device for producing an encoded wideband signal comprising the set of pitch codebook parameters associated to the path having the lowest pitch prediction error, the innovative codebook parameters, and the linear prediction synthesis filter coefficients.

The present invention still further relates to a cellular communication system, a cellular mobile transmitter/receiver unit, a cellular network element, and a bidirectional wireless communication sub-system comprising the above described decoder.

The objects, advantages and other features of the present invention will become more apparent upon reading of the following non restrictive description of a preferred embodiment thereof, given by way of example only with reference to the accompanying drawings.

BRIEF DESCRIPTION OF THE DRAWINGS

In the appended drawings:

5 Figure 1 is a schematic block diagram of a preferred embodiment of wideband encoding device;

10 Figure 2 is a schematic block diagram of a preferred embodiment of wideband decoding device;

15 Figure 3 is a schematic block diagram of a preferred embodiment of pitch analysis device; and

20 Figure 4 is a simplified, schematic block diagram of a cellular communication system in which the wideband encoding device of Figure 1 and the wideband decoding device of Figure 2 can be used.

25 DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

As well known to those of ordinary skill in the art, a cellular communication system such as 401 (see Figure 4) provides a telecommunication service over a large geographic area by dividing that large geographic area into a number C of smaller cells. The C smaller cells

are serviced by respective cellular base stations 402_1 , 402_2 ... 402_c to provide each cell with radio signalling, audio and data channels.

Radio signalling channels are used to page mobile radiotelephones (mobile transmitter/receiver units) such as 403 within the limits of the coverage area (cell) of the cellular base station 402, and to place calls to other radiotelephones 403 located either inside or outside the base station's cell or to another network such as the Public Switched Telephone Network (PSTN) 404.

Once a radiotelephone 403 has successfully placed or received a call, an audio or data channel is established between this radiotelephone 403 and the cellular base station 402 corresponding to the cell in which the radiotelephone 403 is situated, and communication between the base station 402 and radiotelephone 403 is conducted over that audio or data channel. The radiotelephone 403 may also receive control or timing information over a signalling channel while a call is in progress.

If a radiotelephone 403 leaves a cell and enters another adjacent cell while a call is in progress, the radiotelephone 403 hands over the call to an available audio or data channel of the new cell base station 402. If a radiotelephone 403 leaves a cell and enters another adjacent cell while no call is in progress, the radiotelephone 403 sends a control message over the signalling channel to log into the base station 402 of the new cell. In this manner mobile communication over a wide geographical area is possible.

The cellular communication system 401 further comprises a control terminal 405 to control communication between the cellular base stations

402 and the PSTN 404, for example during a communication between a radiotelephone 403 and the PSTN 404, or between a radiotelephone 403 located in a first cell and a radiotelephone 403 situated in a second cell.

Of course, a bidirectional wireless radio communication subsystem
5 is required to establish an audio or data channel between a base station 402 of one cell and a radiotelephone 403 located in that cell. As illustrated in very simplified form in Figure 4, such a bidirectional wireless radio communication subsystem typically comprises in the radiotelephone 403:

- a transmitter 406 including:
10 - an encoder 407 for encoding the voice signal; and
 - a transmission circuit 408 for transmitting the encoded voice signal from the encoder 407 through an antenna such as 409; and
- a receiver 410 including:
15 - a receiving circuit 411 for receiving a transmitted encoded voice signal usually through the same antenna 409; and
 - a decoder 412 for decoding the received encoded voice signal from the receiving circuit 411.

20 The radiotelephone further comprises other conventional radiotelephone circuits 413 to which the encoder 407 and decoder 412 are connected and for processing signals therefrom, which circuits 413 are well known to those of ordinary skill in the art and, accordingly, will not be further described in the present specification.

25

Also, such a bidirectional wireless radio communication subsystem typically comprises in the base station 402:

- a transmitter 414 including:
 - an encoder 415 for encoding the voice signal; and
 - a transmission circuit 416 for transmitting the encoded voice signal from the encoder 415 through an antenna such as 417; and
- 5 - a receiver 418 including:
 - a receiving circuit 419 for receiving a transmitted encoded voice signal through the same antenna 417 or through another antenna (not shown); and
 - 10 - a decoder 420 for decoding the received encoded voice signal from the receiving circuit 419.

The base station 402 further comprises, typically, a base station controller 421, along with its associated database 422, for controlling communication between the control terminal 405 and the transmitter 414 and
15 receiver 418.

As well known to those of ordinary skill in the art, voice encoding is required in order to reduce the bandwidth necessary to transmit sound signal, for example voice signal such as speech, across the bidirectional
20 wireless radio communication subsystem, i.e., between a radiotelephone 403 and a base station 402.

LP voice encoders (such as 415 and 407) typically operating at 13 kbits/second and below such as Code-Excited Linear Prediction (CELP)
25 encoders typically use a LP synthesis filter to model the short-term spectral envelope of the voice signal. The LP information is transmitted, typically,

every 10 or 20 ms to the decoder (such 420 and 412) and is extracted at the decoder end.

The novel techniques disclosed in the present specification may apply to different LP-based coding systems. However, a CELP-type coding system is used in the preferred embodiment for the purpose of presenting a non-limitative illustration of these techniques. In the same manner, such techniques can be used with sound signals other than voice and speech as well with other types of wideband signals.

Figure 1 shows a general block diagram of a CELP-type speech encoding device 100 modified to better accommodate wideband signals.

The sampled input speech signal 114 is divided into successive L -sample blocks called "frames". In each frame, different parameters representing the speech signal in the frame are computed, encoded, and transmitted. LP parameters representing the LP synthesis filter are usually computed once every frame. The frame is further divided into smaller blocks of N samples (blocks of length N), in which excitation parameters (pitch and innovation) are determined. In the CELP literature, these blocks of length N are called "subframes" and the N -sample signals in the subframes are referred to as N -dimensional vectors. In this preferred embodiment, the length N corresponds to 5 ms while the length L corresponds to 20 ms, which means that a frame contains four subframes ($N=80$ at the sampling rate of 16 kHz and 64 after down-sampling to 12.8 kHz). Various N -dimensional vectors occur in the encoding procedure. A list of the vectors which appear in Figures 1 and 2 as well as a list of transmitted parameters are given herein below:

List of the main N -dimensional vectors

- s Wideband signal input speech vector (after down-sampling, pre-processing, and preemphasis);
 5 s_w Weighted speech vector;
 s_o Zero-input response of weighted synthesis filter;
 s_p Down-sampled pre-processed signal;
 Oversampled synthesized speech signal;
 10 s' Synthesis signal before deemphasis;
 s_d Deemphasized synthesis signal;
 s_h Synthesis signal after deemphasis and postprocessing;
 x Target vector for pitch search;
 x' Target vector for innovation search;
 15 h Weighted synthesis filter impulse response;
 v_T Adaptive (pitch) codebook vector at delay T ;
 y_T Filtered pitch codebook vector (v_T convolved with h);
 c_k Innovative codevector at index k (k -th entry from the innovation codebook);
 20 c_f Enhanced scaled innovation codevector;
 u Excitation signal (scaled innovation and pitch codevectors);
 u' Enhanced excitation;
 z Band-pass noise sequence;
 w' White noise sequence; and
 25 w Scaled noise sequence.

List of transmitted parameters

- STP Short term prediction parameters (defining $A(z)$);
- T Pitch lag (or pitch codebook index);
- b Pitch gain (or pitch codebook gain);
- 5 j Index of the low-pass filter used on the pitch codevector;
- k Codevector index (innovation codebook entry); and
- g Innovation codebook gain.

10 In this preferred embodiment, the STP parameters are transmitted once per frame and the rest of the parameters are transmitted four times per frame (every subframe).

ENCODER SIDE

15 The sampled speech signal is encoded on a block by block basis by the encoding device 100 of Figure 1 which is broken down into eleven modules numbered from 101 to 111.

20 The input speech is processed into the above mentioned L -sample blocks called frames.

Referring to Figure 1, the sampled input speech signal 114 is down-sampled in a down-sampling module 101. For example, the signal is down-sampled from 16 kHz down to 12.8 kHz, using techniques well known to those of ordinary skill in the art. Down-sampling down to another frequency can of course be envisaged. Down-sampling increases the coding efficiency, since a smaller frequency bandwidth is encoded. This also

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reduces the algorithmic complexity since the number of samples in a frame is decreased. The use of down-sampling becomes significant when the bit rate is reduced below 16 kbit/s, although down-sampling is not essential above 16 kbit/s.

- 5 After down-sampling, the 320-sample frame of 20 ms is reduced to 256-sample frame (down-sampling ratio of 4/5).

10 The input frame is then supplied to the optional pre-processing block 102. Pre-processing block 102 may consist of a high-pass filter with a 50 Hz cut-off frequency. High-pass filter 102 removes the unwanted sound components below 50 Hz.

15 The down-sampled pre-processed signal is denoted by $s_p(n)$, $n=0, 1, 2, \dots, L-1$, where L is the length of the frame (256 at a sampling frequency of 12.8 kHz). In a preferred embodiment of the preemphasis filter 103, the signal $s_p(n)$ is preemphasized using a filter having the following transfer function:

20
$$P(z) = 1 - \mu z^{-1}$$

25 where μ is a preemphasis factor with a value located between 0 and 1 (a typical value is $\mu = 0.7$). A higher-order filter could also be used. It should be pointed out that high-pass filter 102 and preemphasis filter 103 can be interchanged to obtain more efficient fixed-point implementations.

The function of the preemphasis filter 103 is to enhance the high frequency contents of the input signal. It also reduces the dynamic range of the input speech signal, which renders it more suitable for fixed-point implementation. Without preemphasis, LP analysis in fixed-point using single-precision arithmetic is difficult to implement.

5

Preemphasis also plays an important role in achieving a proper overall perceptual weighting of the quantization error, which contributes to improved sound quality. This will be explained in more detail herein below.

10

The output of the preemphasis filter 103 is denoted $s(n)$. This signal is used for performing LP analysis in calculator module 104. LP analysis is a technique well known to those of ordinary skill in the art. In this preferred embodiment, the autocorrelation approach is used. In the autocorrelation approach, the signal $s(n)$ is first windowed using a Hamming window (having usually a length of the order of 30-40 ms). The autocorrelations are computed from the windowed signal, and Levinson-Durbin recursion is used to compute LP filter coefficients, a_i , where $i=1, \dots, p$, and where p is the LP order, which is typically 16 in wideband coding. The parameters a_i are the coefficients of the transfer function of the LP filter, which is given by the following relation:

15

20

$$A(z) = 1 + \sum_{i=1}^p a_i z^{-i}$$

25

LP analysis is performed in calculator module 104, which also performs the quantization and interpolation of the LP filter coefficients. The LP filter coefficients are first transformed into another equivalent domain more suitable for quantization and interpolation purposes. The line spectral pair (LSP) and immittance spectral pair (ISP) domains are two domains in which quantization and interpolation can be efficiently performed. The 16 LP filter coefficients, a_n , can be quantized in the order of 30 to 50 bits using split or multi-stage quantization, or a combination thereof. The purpose of the interpolation is to enable updating the LP filter coefficients every subframe while transmitting them once every frame, which improves the encoder performance without increasing the bit rate. Quantization and interpolation of the LP filter coefficients is believed to be otherwise well known to those of ordinary skill in the art and, accordingly, will not be further described in the present specification.

The following paragraphs will describe the rest of the coding operations performed on a subframe basis. In the following description, the filter $A(z)$ denotes the unquantized interpolated LP filter of the subframe, and the filter $\hat{A}(z)$ denotes the quantized interpolated LP filter of the subframe.

Perceptual Weighting:

In analysis-by-synthesis encoders, the optimum pitch and innovation parameters are searched by minimizing the mean squared error between the input speech and synthesized speech in a perceptually weighted domain.

This is equivalent to minimizing the error between the weighted input speech and weighted synthesis speech.

The weighted signal $s_w(n)$ is computed in a perceptual weighting filter 105. Traditionally, the weighted signal $s_w(n)$ is computed by a weighting filter having a transfer function $W(z)$ in the form:

$$W(z) = A(z/\gamma_1) / A(z/\gamma_2) \quad \text{where} \quad 0 < \gamma_2 < \gamma_1 \leq 1$$

As well known to those of ordinary skill in the art, in prior art analysis-by-synthesis (AbS) encoders, analysis shows that the quantization error is weighted by a transfer function $W^{-1}(z)$, which is the inverse of the transfer function of the perceptual weighting filter 105. This result is well described by B.S. Atal and M.R. Schroeder in "Predictive coding of speech and subjective error criteria", IEEE Transaction ASSP, vol. 27, no. 3, pp. 247-254, June 1979. Transfer function $W^{-1}(z)$ exhibits some of the formant structure of the input speech signal. Thus, the masking property of the human ear is exploited by shaping the quantization error so that it has more energy in the formant regions where it will be masked by the strong signal energy present in these regions. The amount of weighting is controlled by the factors γ_1 and γ_2 .

20

The above traditional perceptual weighting filter 105 works well with telephone band signals. However, it was found that this traditional perceptual weighting filter 105 is not suitable for efficient perceptual weighting of wideband signals. It was also found that the traditional perceptual weighting filter 105 has inherent limitations in modelling the formant structure and the required spectral tilt concurrently. The spectral tilt is more pronounced in wideband signals due to the wide dynamic range

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between low and high frequencies. The prior art has suggested to add a tilt filter into $W(z)$ in order to control the tilt and formant weighting of the wideband input signal separately.

A novel solution to this problem is, in accordance with the present invention, to introduce the preemphasis filter 103 at the input, compute the LP filter $A(z)$ based on the preemphasized speech $s(n)$, and use a modified filter $W(z)$ by fixing its denominator.

LP analysis is performed in module 104 on the preemphasized signal $s(n)$ to obtain the LP filter $A(z)$. Also, a new perceptual weighting filter 105 with fixed denominator is used. An example of transfer function for the perceptual weighting filter 104 is given by the following relation:

$$W(z) = A(z/\gamma_1) / (1 - \gamma_2 z^{-1}) \quad \text{where} \quad 0 < \gamma_2 < \gamma_1 \leq 1$$

A higher order can be used at the denominator. This structure substantially decouples the formant weighting from the tilt.

20

Note that because $A(z)$ is computed based on the preemphasized speech signal $s(n)$, the tilt of the filter $1/A(z\gamma_1)$ is less pronounced compared to the case when $A(z)$ is computed based on the original speech. Since deemphasis is performed at the decoder end using a filter having the transfer function:

25

$$P^{-1}(z) = 1/(1 - \mu z^{-1}),$$

the quantization error spectrum is shaped by a filter having a transfer function $W^{-1}(z)P^{-1}(z)$. When γ_2 is set equal to μ , which is typically the case, the spectrum of the quantization error is shaped by a filter whose transfer

5 function is $1/A(z\gamma_1)$, with $A(z)$ computed based on the preemphasized speech signal. Subjective listening showed that this structure for achieving the error shaping by a combination of preemphasis and modified weighting filtering is very efficient for encoding wideband signals, in addition to the advantages of ease of fixed-point algorithmic implementation.

Pitch Analysis:

In order to simplify the pitch analysis, an open-loop pitch lag T_{OL} is first estimated in the open-loop pitch search module 106 using the weighted speech signal $s_w(n)$. Then the closed-loop pitch analysis, which is performed in closed-loop pitch search module 107 on a subframe basis, is restricted around the open-loop pitch lag T_{OL} which significantly reduces the search complexity of the LTP parameters T and b (pitch lag and pitch gain). Open-

15 loop pitch analysis is usually performed in module 106 once every 10 ms (two subframes) using techniques well known to those of ordinary skill in the art.

The target vector x for LTP (Long Term Prediction) analysis is first

25 computed. This is usually done by subtracting the zero-input response s_0 of weighted synthesis filter $W(z)/\hat{A}(z)$ from the weighted speech signal $s_w(n)$. This zero-input response s_0 is calculated by a zero-input response calculator

108. More specifically, the target vector x is calculated using the following relation:

$$x = s_w - s_o$$

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where x is the N -dimensional target vector, s_w is the weighted speech vector in the subframe, and s_o is the zero-input response of filter $W(z)/\hat{A}(z)$ which is the output of the combined filter $W(z)/\hat{A}(z)$ due to its initial states.

10 The zero-input response calculator 108 is responsive to the quantized interpolated LP filter $\hat{A}(z)$ from the LP analysis, quantization and interpolation calculator 104 and to the initial states of the weighted synthesis filter $W(z)/\hat{A}(z)$ stored in memory module 111 to calculate the zero-input response s_o (that part of the response due to the initial states as determined by setting
15 the inputs equal to zero) of filter $W(z)/\hat{A}(z)$. This operation is well known to those of ordinary skill in the art and, accordingly, will not be further described.

Of course, alternative but mathematically equivalent approaches can be used to compute the target vector x .

20

A N -dimensional impulse response vector h of the weighted synthesis filter $W(z)/\hat{A}(z)$ is computed in the impulse response generator 109 using the LP filter coefficients $A(z)$ and $\hat{A}(z)$ from module 104. Again, this operation is well known to those of ordinary skill in the art and, accordingly,
25 will not be further described in the present specification.

The closed-loop pitch (or pitch codebook) parameters b , T and j are

computed in the closed-loop pitch search module 107, which uses the target vector $\mathbf{x}$, the impulse response vector $\mathbf{h}$ and the open-loop pitch lag T_{OL} as inputs. Traditionally, the pitch prediction has been represented by a pitch filter having the following transfer function:

5

$$1 / (1 - bz^{-T})$$

10

where b is the pitch gain and T is the pitch delay or lag. In this case, the pitch contribution to the excitation signal $u(n)$ is given by $bu(n-T)$, where the total excitation is given by

15

$$u(n) = bu(n-T) + gc_k(n)$$

with g being the innovative codebook gain and $c_k(n)$ the innovative codevector at index k .

20

This representation has limitations if the pitch lag T is shorter than the subframe length N . In another representation, the pitch contribution can be seen as a pitch codebook containing the past excitation signal. Generally, each vector in the pitch codebook is a shift-by-one version of the previous vector (discarding one sample and adding a new sample). For pitch lags

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$T > N$, the pitch codebook is equivalent to the filter structure $(1/(1-bz^{-T}))$, and a pitch codebook vector $v_T(n)$ at pitch lag T is given by

$$v_T(n) = u(n-T) \quad , \quad n=0, \dots, N-1.$$

For pitch lags T shorter than N , a vector $v_T(n)$ is built by repeating the available samples from the past excitation until the vector is completed (this is not equivalent to the filter structure).

In recent encoders, a higher pitch resolution is used which significantly improves the quality of voiced sound segments. This is achieved by oversampling the past excitation signal using polyphase interpolation filters. In this case, the vector $v_T(n)$ usually corresponds to an interpolated version of the past excitation, with pitch lag T being a non-integer delay (e.g. 50.25).

The pitch search consists of finding the best pitch lag T and gain b that minimize the mean squared weighted error E between the target vector x and the scaled filtered past excitation. Error E being expressed as:

$$E = \|x - by_T\|^2$$

where y_T is the filtered pitch codebook vector at pitch lag T :

$$y_T(n) = v_T(n) * h(n) = \sum_{i=0}^n v_T(i)h(n-i) \quad , \quad n=0, \dots, N-1.$$

It can be shown that the error E is minimized by maximizing the search criterion

5

$$C = \frac{x^t y_T}{\sqrt{y^t y_T}}$$

10

where t denotes vector transpose.

In the preferred embodiment of the present invention, a 1/3 subsample pitch resolution is used, and the pitch (pitch codebook) search is composed of three stages.

15

In the first stage, an open-loop pitch lag T_{OL} is estimated in open-loop pitch search module 106 in response to the weighted speech signal $s_w(n)$. As indicated in the foregoing description, this open-loop pitch analysis is usually performed once every 10 ms (two subframes) using techniques well known to those of ordinary skill in the art.

20

In the second stage, the search criterion C is searched in the closed-loop pitch search module 107 for integer pitch lags around the estimated open-loop pitch lag T_{OL} (usually ± 5), which significantly simplifies the search procedure. A simple procedure is used for updating the filtered codevector y_T without the need to compute the convolution for every pitch lag.

25

Once an optimum integer pitch lag is found in the second stage, a third stage of the search (module 107) tests the fractions around that optimum integer pitch lag.

When the pitch predictor is represented by a filter of the form $1/(1-bz^{-T})$, which is a valid assumption for pitch lags $T > N$, the spectrum of the pitch filter exhibits a harmonic structure over the entire frequency range, with a harmonic frequency related to $1/T$. In case of wideband signals, this structure is not very efficient since the harmonic structure in wideband signals does not cover the entire extended spectrum. The harmonic structure exists only up to a certain frequency, depending on the speech segment. Thus, in order to achieve efficient representation of the pitch contribution in voiced segments of wideband speech, the pitch prediction filter needs to have the flexibility of varying the amount of periodicity over the wideband spectrum.

A new method which achieves efficient modeling of the harmonic structure of the speech spectrum of wideband signals is disclosed in the present specification, whereby several forms of low pass filters are applied to the past excitation and the low pass filter with higher prediction gain is selected.

When subsample pitch resolution is used, the low pass filters can be incorporated into the interpolation filters used to obtain the higher pitch resolution. In this case, the third stage of the pitch search, in which the fractions around the chosen integer pitch lag are tested, is repeated for the several interpolation filters having different low-pass characteristics and the fraction and filter index which maximize the search criterion C are selected.

A simpler approach is to complete the search in the three stages described above to determine the optimum fractional pitch lag using only one interpolation filter with a certain frequency response, and select the optimum low-pass filter shape at the end by applying the different predetermined low-pass filters to the chosen pitch codebook vector $\mathbf{v}_T$ and select the low-pass filter which minimizes the pitch prediction error. This approach is discussed in detail below.

Figure 3 illustrates a schematic block diagram of a preferred embodiment of the proposed approach.

In memory module 303, the past excitation signal $u(n)$, $n < 0$, is stored. The pitch codebook search module 301 is responsive to the target vector $\mathbf{x}$, to the open-loop pitch lag T_{OL} and to the past excitation signal $u(n)$, $n < 0$, from memory module 303 to conduct a pitch codebook (pitch codebook) search minimizing the above-defined search criterion C . From the result of the search conducted in module 301, module 302 generates the optimum pitch codebook vector $\mathbf{v}_T$. Note that since a sub-sample pitch resolution is used (fractional pitch), the past excitation signal $u(n)$, $n < 0$, is interpolated and the pitch codebook vector $\mathbf{v}_T$ corresponds to the interpolated past excitation signal. In this preferred embodiment, the interpolation filter (in module 301, but not shown) has a low-pass filter characteristic removing the frequency contents above 7000 Hz.

In a preferred embodiment, K filter characteristics are used; these filter characteristics could be low-pass or band-pass filter characteristics. Once the optimum codevector $\mathbf{v}_T$ is determined and supplied by the pitch codevector generator 302, K filtered versions of $\mathbf{v}_T$ are computed

respectively using K different frequency shaping filters such as 305^(j), where $j=1, 2, \dots, K$. These filtered versions are denoted $v_f^{(j)}$, where $j=1, 2, \dots, K$. The different vectors $v_f^{(j)}$ are convolved in respective modules 304^(j), where $j=0, 1, 2, \dots, K$, with the impulse response h to obtain the vectors $y^{(j)}$, where $j=0, 1, 2, \dots, K$. To calculate the mean squared pitch prediction error for each vector $y^{(j)}$, the value $y^{(j)}$ is multiplied by the gain b by means of a corresponding amplifier 307^(j) and the value $by^{(j)}$ is subtracted from the target vector x by means of a corresponding subtractor 308^(j). Selector 309 selects the frequency shaping filter 305^(j) which minimizes the mean squared pitch prediction error

$$e^{(j)} = \|x - b^{(j)} y^{(j)}\|^2, \quad j=1, 2, \dots, K$$

To calculate the mean squared pitch prediction error $e^{(j)}$ for each value of $y^{(j)}$, the value $y^{(j)}$ is multiplied by the gain b by means of a corresponding amplifier 307^(j) and the value $b^{(j)} y^{(j)}$ is subtracted from the target vector x by means of subtractors 308^(j). Each gain $b^{(j)}$ is calculated in a corresponding gain calculator 306^(j) in association with the frequency shaping filter at index j , using the following relationship:

$$b^{(j)} = x^T y^{(j)} / \|y^{(j)}\|^2$$

In selector 309, the parameters b , T , and j are chosen based on v_T or $v_f^{(j)}$ which minimizes the mean squared pitch prediction error e .

Referring back to Figure 1, the pitch codebook index T is encoded and transmitted to multiplexer 112. The pitch gain b is quantized and transmitted to multiplexer 112. With this new approach, extra information is needed to encode the index j of the selected frequency shaping filter in multiplexer 112. For example, if three filters are used ($j=0, 1, 2, 3$), then two bits are needed to represent this information. The filter index information j can also be encoded jointly with the pitch gain b .

Innovative codebook search:

Once the pitch, or LTP (Long Term Prediction) parameters b , T , and j are determined, the next step is to search for the optimum innovative excitation by means of search module 110 of Figure 1. First, the target vector x is updated by subtracting the LTP contribution:

$$x' = x - by_T$$

where b is the pitch gain and y_T is the filtered pitch codebook vector (the past excitation at delay T filtered with the selected low pass filter and convolved with the impulse response h as described with reference to Figure 3).

The search procedure in CELP is performed by finding the optimum excitation codevector c_k and gain g which minimize the mean-squared error between the target vector and the scaled filtered codevector

$$E = \| \mathbf{x}' - gH\mathbf{c}_k \|^2$$

where H is a lower triangular convolution matrix derived from the impulse
 5 response vector $\mathbf{h}$.

In the preferred embodiment of the present invention, the innovative
 codebook search is performed in module 110 by means of an algebraic
 codebook as described in US patents Nos: 5,444,816 (Adoul et al.) issued
 10 on August 22, 1995; 5,699,482 granted to Adoul et al., on December 17,
 1997; 5,754,976 granted to Adoul et al., on May 19, 1998; and 5,701,392
 (Adoul et al.) dated December 23, 1997.

Once the optimum excitation codevector $\mathbf{c}_k$ and its gain g are chosen
 15 by module 110, the codebook index k and gain g are encoded and
 transmitted to multiplexer 112.

Referring to Figure 1, the parameters b , T , j , $\hat{A}(z)$, k and g are
 multiplexed through the multiplexer 112 before being transmitted through a
 20 communication channel.

Memory update:

25 In memory module 111 (Figure 1), the states of the weighted
 synthesis filter $W(z)/\hat{A}(z)$ are updated by filtering the excitation signal

$u = gc_k + bv_T$ through the weighted synthesis filter. After this filtering, the states of the filter are memorized and used in the next subframe as initial states for computing the zero-input response in calculator module 108.

As in the case of the target vector x , other alternative but mathematically equivalent approaches well known to those of ordinary skill in the art can be used to update the filter states.

DECODER SIDE

The speech decoding device 200 of Figure 2 illustrates the various steps carried out between the digital input 222 (input stream to the demultiplexer 217) and the output sampled speech 223 (output of the adder 221).

Demultiplexer 217 extracts the synthesis model parameters from the binary information received from a digital input channel. From each received binary frame, the extracted parameters are:

- the short-term prediction parameters (STP) $\hat{A}(z)$ (once per frame);
- the long-term prediction (LTP) parameters T , b , and j (for each subframe); and
- the innovation codebook index k and gain g (for each subframe).

The current speech signal is synthesized based on these parameters as will be explained hereinbelow.

5 The innovative codebook 218 is responsive to the index k to produce the innovation codevector $\mathbf{c}_k$, which is scaled by the decoded gain factor g through an amplifier 224. In the preferred embodiment, an innovative codebook 218 as described in the above mentioned US patent numbers 5,444,816; 5,699,482; 5,754,976; and 5,701,392 is used to represent the innovative codevector $\mathbf{c}_k$.

10 The generated scaled codevector $g\mathbf{c}_k$ at the output of the amplifier 224 is processed through a innovation filter 205.

Periodicity enhancement:

15 The generated scaled codevector at the output of the amplifier 224 is processed through a frequency-dependent pitch enhancer 205.

Enhancing the periodicity of the excitation signal $\mathbf{u}$ improves the quality in case of voiced segments. This was done in the past by filtering
 20 the innovation vector from the innovative codebook (fixed codebook) 218 through a filter in the form $1/(1-\varepsilon bz^{-T})$ where ε is a factor below 0.5 which controls the amount of introduced periodicity. This approach is less efficient in case of wideband signals since it introduces periodicity over the entire spectrum. A new alternative approach, which is part of the
 25 present invention, is disclosed whereby periodicity enhancement is achieved by filtering the innovative codevector $\mathbf{c}_k$ from the innovative (fixed) codebook through an innovation filter 205 ($F(z)$) whose frequency

response emphasizes the higher frequencies more than lower frequencies. The coefficients of $F(z)$ are related to the amount of periodicity in the excitation signal u .

Many methods known to those skilled in the art are available for obtaining valid periodicity coefficients. For example, the value of gain b provides an indication of periodicity. That is, if gain b is close to 1, the periodicity of the excitation signal u is high, and if gain b is less than 0.5, then periodicity is low.

Another efficient way to derive the filter $F(z)$ coefficients used in a preferred embodiment, is to relate them to the amount of pitch contribution in the total excitation signal u . This results in a frequency response depending on the subframe periodicity, where higher frequencies are more strongly emphasized (stronger overall slope) for higher pitch gains. Innovation filter 205 has the effect of lowering the energy of the innovative codevector c_k at low frequencies when the excitation signal u is more periodic, which enhances the periodicity of the excitation signal u at lower frequencies more than higher frequencies. Suggested forms for innovation filter 205 are

$$(1) F(z) = 1 - \sigma z^{-1}, \quad \text{or} \quad (2) F(z) = -\alpha z + 1 - \alpha z^{-1}$$

where σ or α are periodicity factors derived from the level of periodicity of the excitation signal u .

The second three-term form of $F(z)$ is used in a preferred embodiment. The periodicity factor α is computed in the voicing factor generator 204. Several methods can be used to derive the periodicity factor α based on the periodicity of the excitation signal u . Two methods are presented below.

5

Method 1:

The ratio of pitch contribution to the total excitation signal u is first computed in voicing factor generator 204 by

10

$$R_p = \frac{b^2 v_T^t v_T}{u^t u} = \frac{b^2 \sum_{n=0}^{N-1} v_T^2(n)}{\sum_{n=0}^{N-1} u^2(n)}$$

15

where v_T is the pitch codebook vector, b is the pitch gain, and u is the excitation signal u given at the output of the adder 219 by

20

$$u = gc_k + bv_T$$

25

Note that the term bv_T has its source in the pitch codebook (pitch codebook) 201 in response to the pitch lag T and the past value of u stored in memory 203. The pitch codevector v_T from the pitch codebook 201 is then processed through a low-pass filter 202 whose cut-off frequency is adjusted by means of the index j from the demultiplexer 217. The resulting codevector v_T is then multiplied by the gain b from the

demultiplexer 217 through an amplifier 226 to obtain the signal bv_T .

The factor α is calculated in voicing factor generator 204 by

$$\alpha = qR_p \quad \text{bounded by} \quad \alpha < q$$

where q is a factor which controls the amount of enhancement (q is set to 0.25 in this preferred embodiment).

Method 2:

Another method used in a preferred embodiment of the invention for calculating periodicity factor α is discussed below.

First, a voicing factor r_v is computed in voicing factor generator 204 by

$$r_v = (E_v - E_c) / (E_v + E_c)$$

where E_v is the energy of the scaled pitch codevector bv_T and E_c is the energy of the scaled innovative codevector gc_k . That is

$$E_v = b^2 v_T^t v_T = b^2 \sum_{n=0}^{N-1} v_T^2(n)$$

and

$$E_c = g^2 c_k^t c_k = g^2 \sum_{n=0}^{N-1} c_k^2(n)$$

Note that the value of r_v lies between -1 and 1 (1 corresponds to purely voiced signals and -1 corresponds to purely unvoiced signals).

In this preferred embodiment, the factor α is then computed in voicing factor generator 204 by

5

$$\alpha = 0.125 (1 + r_v)$$

10 which corresponds to a value of 0 for purely unvoiced signals and 0.25 for purely voiced signals.

In the first, two-term form of $F(z)$, the periodicity factor σ can be approximated by using $\sigma = 2\alpha$ in methods 1 and 2 above. In such a case, the periodicity factor σ is calculated as follows in method 1 above:

15 $\sigma = 2qR_p$ bounded by $\sigma < 2q$.

In method 2, the periodicity factor σ is calculated as follows:

20 $\sigma = 0.25 (1 + r_v)$.

The enhanced signal c_r is therefore computed by filtering the scaled innovative codevector gc_k through the innovation filter 205 ($F(z)$).

The enhanced excitation signal u' is computed by the adder 220

25 as:

$$u' = c_T + bv_T$$

Note that this process is not performed at the encoder 100. Thus, it is essential to update the content of the pitch codebook 201 using the excitation signal u without enhancement to keep synchronism between the encoder 100 and decoder 200. Therefore, the excitation signal u is used to update the memory 203 of the pitch codebook 201 and the enhanced excitation signal u' is used at the input of the LP synthesis filter 206.

Synthesis and deemphasis

The synthesized signal s' is computed by filtering the enhanced excitation signal u' through the LP synthesis filter 206 which has the form $1/\hat{A}(z)$, where $\hat{A}(z)$ is the interpolated LP filter in the current subframe. As can be seen in Figure 2, the quantized LP coefficients $\hat{A}(z)$ on line 225 from demultiplexer 217 are supplied to the LP synthesis filter 206 to adjust the parameters of the LP synthesis filter 206 accordingly. The deemphasis filter 207 is the inverse of the preemphasis filter 103 of Figure 1. The transfer function of the deemphasis filter 207 is given by

$$D(z) = 1 / (1 - \mu z^{-1})$$

where μ is a preemphasis factor with a value located between 0 and 1 (a typical value is $\mu = 0.7$). A higher-order filter could also be used.

The vector $\mathbf{s}'$ is filtered through the deemphasis filter $D(z)$ (module 207) to obtain the vector $\mathbf{s}_b$ which is passed through the high-pass filter 208 to remove the unwanted frequencies below 50 Hz and further obtain $\mathbf{s}_h$.

Oversampling and high-frequency regeneration

The over-sampling module 209 conducts the inverse process of the down-sampling module 101 of Figure 1. In this preferred embodiment, oversampling converts from the 12.8 kHz sampling rate to the original 16 kHz sampling rate, using techniques well known to those of ordinary skill in the art. The oversampled synthesis signal is denoted $\hat{\mathbf{s}}$. Signal $\hat{\mathbf{s}}$ is also referred to as the synthesized wideband intermediate signal.

The oversampled synthesis $\hat{\mathbf{s}}$ signal does not contain the higher frequency components which were lost by the downsampling process (module 101 of Figure 1) at the encoder 100. This gives a low-pass perception to the synthesized speech signal. To restore the full band of the original signal, a high frequency generation procedure is disclosed. This procedure is performed in modules 210 to 216, and adder 221, and requires input from voicing factor generator 204 (Figure 2).

In this new approach, the high frequency contents are generated by filling the upper part of the spectrum with a white noise properly scaled in the excitation domain, then converted to the speech domain, preferably by

shaping it with the same LP synthesis filter used for synthesizing the down-sampled signal $\hat{S}$.

The high frequency generation procedure in accordance with the present invention is described hereinbelow.

5

The random noise generator 213 generates a white noise sequence w' with a flat spectrum over the entire frequency bandwidth, using techniques well known to those of ordinary skill in the art. The generated sequence is of length N' which is the subframe length in the original domain. Note that N is the subframe length in the down-sampled domain. In this preferred embodiment, $N=64$ and $N'=80$ which correspond to 5 ms.

10

The white noise sequence is properly scaled in the gain adjusting module 214. Gain adjustment comprises the following steps. First, the energy of the generated noise sequence w' is set equal to the energy of the enhanced excitation signal u' computed by an energy computing module 210, and the resulting scaled noise sequence is given by

15

20

$$w(n) = w'(n) \sqrt{\frac{\sum_{n=0}^{N-1} u'^2(n)}{\sum_{n=0}^{N'-1} w'^2(n)}}, \quad n=0, \dots, N'-1.$$

25

The second step in the gain scaling is to take into account the high frequency contents of the synthesized signal at the output of the voicing factor generator 204 so as to reduce the energy of the generated noise in

case of voiced segments (where less energy is present at high frequencies compared to unvoiced segments). In this preferred embodiment, measuring the high frequency contents is implemented by measuring the tilt of the synthesis signal through a spectral tilt calculator 212 and reducing the energy accordingly. Other measurements such as zero crossing

5 measurements can equally be used. When the tilt is very strong, which corresponds to voiced segments, the noise energy is further reduced. The tilt factor is computed in module 212 as the first correlation coefficient of the synthesis signal s_h and it is given by:

$$tilt = \frac{\sum_{n=1}^{N-1} s_h(n) s_h(n-1)}{\sum_{n=0}^{N-1} s_h^2(n)} \quad , \quad \text{conditioned by } tilt \geq 0 \text{ and } tilt \geq r_v$$

where voicing factor r_v is given by

$$r_v = (E_v - E_c) / (E_v + E_c)$$

25 where E_v is the energy of the scaled pitch codevector bv_T and E_c is the energy of the scaled innovative codevector gc_k , as described earlier. Voicing factor r_v is most often less than $tilt$ but this condition was introduced as a

precaution against high frequency tones where the tilt value is negative and the value of r_v is high. Therefore, this condition reduces the noise energy for such tonal signals.

The tilt value is 0 in case of flat spectrum and 1 in case of strongly voiced signals, and it is negative in case of unvoiced signals where more energy is present at high frequencies.

Different methods can be used to derive the scaling factor g_t from the amount of high frequency contents. In this invention, two methods are given based on the tilt of signal described above.

Method 1:

The scaling factor g_t is derived from the tilt by

$$g_t = 1 - \text{tilt} \quad \text{bounded by} \quad 0.2 \leq g_t \leq 1.0$$

For strongly voiced signal where the tilt approaches 1, g_t is 0.2 and for strongly unvoiced signals g_t becomes 1.0.

Method 2:

The tilt factor g_t is first restricted to be larger or equal to zero, then the scaling factor is derived from the tilt by

$$g_t = 10^{-0.6 \text{tilt}}$$

The scaled noise sequence w_g produced in gain adjusting module 214 is therefore given by:

5

$$w_g = g_t w.$$

When the tilt is close to zero, the scaling factor g_t is close to 1, which does not result in energy reduction. When the tilt value is 1, the scaling factor g_t results in a reduction of 12 dB in the energy of the generated noise.

10

Once the noise is properly scaled (w_g), it is brought into the speech domain using the spectral shaper 215. In the preferred embodiment, this is achieved by filtering the noise w_g through a bandwidth expanded version of the same LP synthesis filter used in the down-sampled domain ($1/\hat{A}(z/0.8)$). The corresponding bandwidth expanded LP filter coefficients are calculated in spectral shaper 215.

15

The filtered scaled noise sequence w_f is then band-pass filtered to the required frequency range to be restored using the band-pass filter 216. In the preferred embodiment, the band-pass filter 216 restricts the noise sequence to the frequency range 5.6-7.2 kHz. The resulting band-pass filtered noise sequence z is added in adder 221 to the oversampled synthesized speech signal $\hat{s}$ to obtain the final reconstructed sound signal s_{out} on the output 223.

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Although the present invention has been described hereinabove by way of a preferred embodiment thereof, this embodiment can be modified at will, within the scope of the appended claims, without departing from the spirit and nature of the subject invention. Even though the preferred embodiment discusses the use of wideband speech signals, it will be
5 obvious to those skilled in the art that the subject invention is also directed to other embodiments using wideband signals in general and that it is not necessarily limited to speech applications.

PCT/CA99/01008

WHAT IS CLAIMED IS:

1. A pitch analysis device for producing an optimal set of pitch codebook parameters, comprising:

5 a) at least two signal paths associated to respective sets of pitch codebook parameters, wherein:

i) each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device; and

10 ii) at least one of said two paths comprises a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of said one path; and

15 b) a selector for comparing the pitch prediction errors calculated in said at least two signal paths, for choosing the signal path having the lowest calculated pitch prediction error, and for selecting the set of pitch codebook parameters associated to the chosen signal path.

20 2. A pitch analysis device as defined in claim 1, wherein one of said at least two paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

25 3. A pitch analysis device as defined in claim 1, wherein said signal paths comprises a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same path.

4. A pitch analysis device as defined in claim 3, wherein the filters of said plurality of paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

5. A pitch analysis device as defined in claim 1, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with a weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

6. A pitch analysis device as defined in claim 5, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = \frac{x^T y^{(j)}}{\|y^{(j)}\|^2} \quad \text{where } j = 0, 1, 2, \dots, K, \text{ and } K \text{ corresponds to a number of signal paths,}$$

and where x is said pitch search target vector, and $y^{(j)}$ is said convolved pitch codevector.

7. A pitch analysis device as defined in claim 1, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the
5 signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.
8. A pitch analysis device as defined in claim 5, wherein:
- a) each of said filters of the plurality of signal paths is identified by a filter
10 index;
 - b) said pitch codevector is identified by a pitch codebook index; and
 - c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.
9. A pitch analysis device as defined in claim 1, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.
10. A pitch analysis method for producing an optimal set of pitch codebook parameters, comprising:
- a) in at least two signal paths associated to respective sets of pitch codebook parameters, calculating, for each signal path, a pitch prediction error of a pitch codevector from a pitch codebook search device;
 - 25 b) in at least one of said two signal paths, filtering the pitch codevector before supplying said pitch codevector for calculation of said pitch prediction error of said one path; and

c) comparing the pitch prediction errors calculated in said at least two signal paths, choosing the signal path having the lowest calculated pitch prediction error, and selecting the set of pitch codebook parameters associated to the chosen signal path.

5 11. A pitch analysis method as defined in claim 10, wherein, in one of said at least two paths, no filtering of the pitch codevector is performed before supplying said pitch codevector to the pitch prediction error calculating device.

10 12. A pitch analysis method as defined in claim 10, wherein said signal paths comprises a plurality of signal paths and wherein filtering the pitch codevector is performed in each of said plurality of signal paths before supplying said pitch codevector to the pitch prediction error calculating device of the same path.

15 13. A pitch analysis method as defined in claim 12, further comprising selecting the filters of said plurality of paths from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

20 14. A pitch analysis method as defined in claim 10, wherein calculating a pitch prediction error in each signal path comprises:

25 a) convolving the pitch codevector with a weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;

c) multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

- 5 15. A pitch analysis method as defined in claim 14, wherein said pitch gain calculation comprises calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = \frac{x^T y^{(j)}}{\|y^{(j)}\|^2} \quad \text{where } j = 0, 1, 2, \dots, K, \text{ and } K \text{ corresponds to a number of signal paths,}$$

10

and where x is said pitch search target vector, and $y^{(j)}$ is said convolved pitch codevector.

- 15 16. A pitch analysis method as defined in claim 10, wherein calculating said pitch prediction error, in each signal path, comprises calculating an energy of the corresponding pitch prediction error, and wherein comparing the pitch prediction error comprises comparing the energies of said pitch prediction errors of the different signal paths and choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest
20 calculated energy of the pitch prediction error.

17. A pitch analysis method as defined in claim 14, wherein:

- a) identifying each of said filters of the plurality of signal paths by a filter index;
- 25 b) identifying said pitch codevector by a pitch codebook index; and
- c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

18. A pitch analysis method as defined in claim 10, wherein said filtering the pitch codevector is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

- 5 19. An encoder having a pitch analysis device as in claim 1 for encoding a wideband input signal, said encoder comprising:
- a) a linear prediction synthesis filter calculator responsive to the wideband signal for producing linear prediction synthesis filter coefficients;
 - b) a perceptual weighting filter, responsive to the wideband signal
10 and the linear prediction synthesis filter coefficients, for producing a perceptually weighted signal;
 - c) an impulse response generator responsive to said linear prediction synthesis filter coefficients for producing a weighted synthesis filter impulse response signal;
 - 15 d) a pitch search unit for producing pitch codebook parameters, said pitch search unit comprising:
 - i) said pitch codebook search device responsive to the perceptually weighted signal and the linear prediction synthesis filter coefficients for producing the pitch codevector and an
20 innovative search target vector; and
 - ii) said pitch analysis device responsive to the pitch codevector for selecting, from said sets of pitch codebook parameters, the set of pitch codebook parameters associated to the path having the lowest calculated pitch prediction error;
 - 25 d) an innovative codebook search device, responsive to the weighted synthesis filter impulse response signal, and the innovative search target vector, for producing innovative codebook parameters; and

e) a signal forming device for producing an encoded wideband signal comprising the set of pitch codebook parameters associated to the path having the lowest pitch prediction error, said innovative codebook parameters, and said linear prediction synthesis filter coefficients.

- 5 20. An encoder as defined in claim 19, wherein one of said at least two paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.
- 10 21. An encoder as defined in claim 19, wherein said signal paths comprises a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same path.
- 15 22. An encoder as defined in claim 21, wherein the filters of said plurality of paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.
23. An encoder as defined in claim 19, wherein each pitch prediction error calculating device comprises:
- 20 a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;
- b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and the pitch search target vector;
- 25 c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

24. An encoder as defined in claim 23, wherein said pitch gain calculator
5 comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^T y^{(j)} / \|y^{(j)}\|^2 \quad \text{where } j = 0, 1, 2, \dots, K, \text{ and } K \text{ corresponds to a number of signal paths,}$$

- 10 and where x is said pitch search target vector, and $y^{(j)}$ is said convolved pitch codevector.

25. An encoder as defined in claim 19, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector
15 comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

20

26. An encoder as defined in claim 23, wherein:

a) each of said filters of the plurality of signal paths is identified by a filter index;

b) said pitch codevector is identified by a pitch codebook index; and

- 25 c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

27. A encoder as defined in claim 19, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

28. A cellular communication system for servicing a large geographical area
5 divided into a plurality of cells, comprising:

a) mobile transmitter/receiver units;

b) cellular base stations respectively situated in said cells;

c) a control terminal for controlling communication between the cellular
base stations;

10 d) a bidirectional wireless communication sub-system between each
mobile unit situated in one cell and the cellular base station of said one cell,
said bidirectional wireless communication sub-system comprising, in both the
mobile unit and the cellular base station:

15 i) a transmitter including an encoder for encoding a
wideband signal as recited in claim 19 and a transmission circuit
for transmitting the encoded wideband signal; and

ii) a receiver including a receiving circuit for receiving a
transmitted encoded wideband signal and a decoder for decoding
the received encoded wideband signal.

20

29. A cellular communication system as defined in claim 28, wherein one of
said at least two paths comprises no filter for filtering the pitch codevector
before supplying said pitch codevector to the pitch prediction error calculating
device.

25

30. A cellular communication system as defined in claim 28, wherein said
signal paths comprises a plurality of signal paths each provided with a filter

for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same path.

31. A cellular communication system as defined in claim 30, wherein the filters of said plurality of paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

32. A cellular communication system as defined in claim 28, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and the pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

33. A cellular communication system as defined in claim 32, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^{(j)} y^{(j)} / \|y^{(j)}\|^2 \quad \text{where } j = 0, 1, 2, \dots, K, \text{ and } K \text{ corresponds to a number of signal paths,}$$

and where x is said pitch search target vector, and $y^{(j)}$ is said convolved pitch codevector.

34. A cellular communication system as defined in claim 28, wherein said pitch prediction error calculating device of each signal path comprises
5 means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

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35. A cellular communication system as defined in claim 32, wherein:

a) each of said filters of the plurality of signal paths is identified by a filter index;

b) said pitch codevector is identified by a pitch codebook index; and

15

c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

20

36. A cellular communication system as defined in claim 28, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

25

37. A cellular mobile transmitter/receiver unit comprising:

a) a transmitter including an encoder for encoding a wideband signal as recited in claim 19 and a transmission circuit for transmitting the encoded wideband signal; and

b) a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal.

38. A cellular mobile transmitter/receiver unit as defined in claim 37, wherein
5 one of said at least two paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

39. A cellular mobile transmitter/receiver unit as defined in claim 37, wherein
10 said signal paths comprises a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same path.

40. A cellular mobile transmitter/receiver unit as defined in claim 39, wherein
15 the filters of said plurality of paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

41. A cellular mobile transmitter/receiver unit as defined in claim 37, wherein
20 each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response
25 to the convolved pitch codevector and the pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

42. A cellular mobile transmitter/receiver unit as defined in claim 41, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^T y^{(j)} / \|y^{(j)}\|^2 \quad \text{where } j = 0, 1, 2, \dots, K, \text{ and } K \text{ corresponds to a number of signal paths,}$$

and where x is said pitch search target vector, and $y^{(j)}$ is said convolved pitch codevector.

43. A cellular mobile transmitter/receiver unit as defined in claim 37, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

44. A cellular mobile transmitter/receiver unit as defined in claim 41, wherein:

a) each of said filters of the plurality of signal paths is identified by a filter index;

b) said pitch codevector is identified by a pitch codebook index; and

c) said pitch codebook parameters comprise the filter index, the
5 pitch codebook index and the pitch gain.

45. A cellular mobile transmitter/receiver unit as defined in claim 37, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version
10 of said pitch codevector.

46. A cellular network element comprising:
a) a transmitter including an encoder for encoding a wideband signal as recited in claim 19 and a transmission circuit for transmitting the encoded wideband signal; and
15

b) a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal.

20 47. A cellular network element as defined in claim 46, wherein one of said at least two paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

25 48. A cellular network element as defined in claim 46, wherein said signal paths comprises a plurality of signal paths each provided with a filter for

filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same path.

49. A cellular network element as defined in claim 48, wherein the filters of said plurality of paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

50. A cellular network element as defined in claim 46, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and the pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

51. A cellular network element as defined in claim 50, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^T y^{(j)} / \|y^{(j)}\|^2 \quad \text{where } j = 0, 1, 2, \dots, K, \text{ and } K \text{ corresponds to a number of signal paths,}$$

and where x is said pitch search target vector, and $y^{(j)}$ is said convolved pitch codevector.

52. A cellular network element as defined in claim 46, wherein said pitch
5 prediction error calculating device of each signal path comprises means for
calculating an energy of the corresponding pitch prediction error, and
wherein said selector comprises means for comparing the energies of said
pitch prediction errors of the different signal paths and for choosing as the
10 signal path having the lowest calculated pitch prediction error the signal path
having the lowest calculated energy of the pitch prediction error.

53. A cellular network element as defined in claim 50, wherein:
a) each of said filters of the plurality of signal paths is identified by
a filter index;
15 b) said pitch codevector is identified by a pitch codebook index;
and
c) said pitch codebook parameters comprise the filter index, the
pitch codebook index and the pitch gain.

20 54. A cellular network element as defined in claim 46, wherein said filter is
integrated in an interpolation filter of said pitch codebook search device, said
interpolation filter being used to produce a sub-sample version of said pitch
codevector.

25 55. In a cellular communication system for servicing a large geographical
area divided into a plurality of cells, comprising: mobile transmitter/receiver

units; cellular base stations, respectively situated in said cells; and control terminal for controlling communication between the cellular base stations:

a bidirectional wireless communication sub-system between each mobile unit situated in one cell and the cellular base station of said one cell, said bidirectional wireless communication sub-system comprising, in both the

5 mobile unit and the cellular base station:

a) a transmitter including an encoder for encoding a wideband signal as recited in claim 19 and a transmission circuit for transmitting the encoded wideband signal; and

b) a receiver including a receiving circuit for receiving a transmitted

10 encoded wideband signal and a decoder for decoding the received encoded wideband signal.

56. A bidirectional wireless communication sub-system as defined in claim 55, wherein one of said at least two paths comprises no filter for filtering the

15 pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

57. A bidirectional wireless communication sub-system as defined in claim 55, wherein said signal paths comprises a plurality of signal paths each

20 provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same path.

58. A bidirectional wireless communication sub-system as defined in claim

25 57, wherein the filters of said plurality of paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

59. A bidirectional wireless communication sub-system as defined in claim 55, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

5 b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and the pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

10 d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

60. A bidirectional wireless communication sub-system as defined in claim 15 59, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = \mathbf{x}^T \mathbf{y}^{(j)} / \|\mathbf{y}^{(j)}\|^2 \quad \text{where } j = 0, 1, 2, \dots, K, \text{ and } K \text{ corresponds to a number of signal paths,}$$

20

and where $\mathbf{x}$ is said pitch search target vector, and $\mathbf{y}^{(j)}$ is said convolved pitch codevector.

61. A bidirectional wireless communication sub-system as defined in claim 25 55, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing

the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

5 62. A bidirectional wireless communication sub-system as defined in claim 59, wherein:

a) each of said filters of the plurality of signal paths is identified by a filter index;

b) said pitch codevector is identified by a pitch codebook index;

10 and

c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

15 63. A bidirectional wireless communication sub-system as defined in claim 55, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.



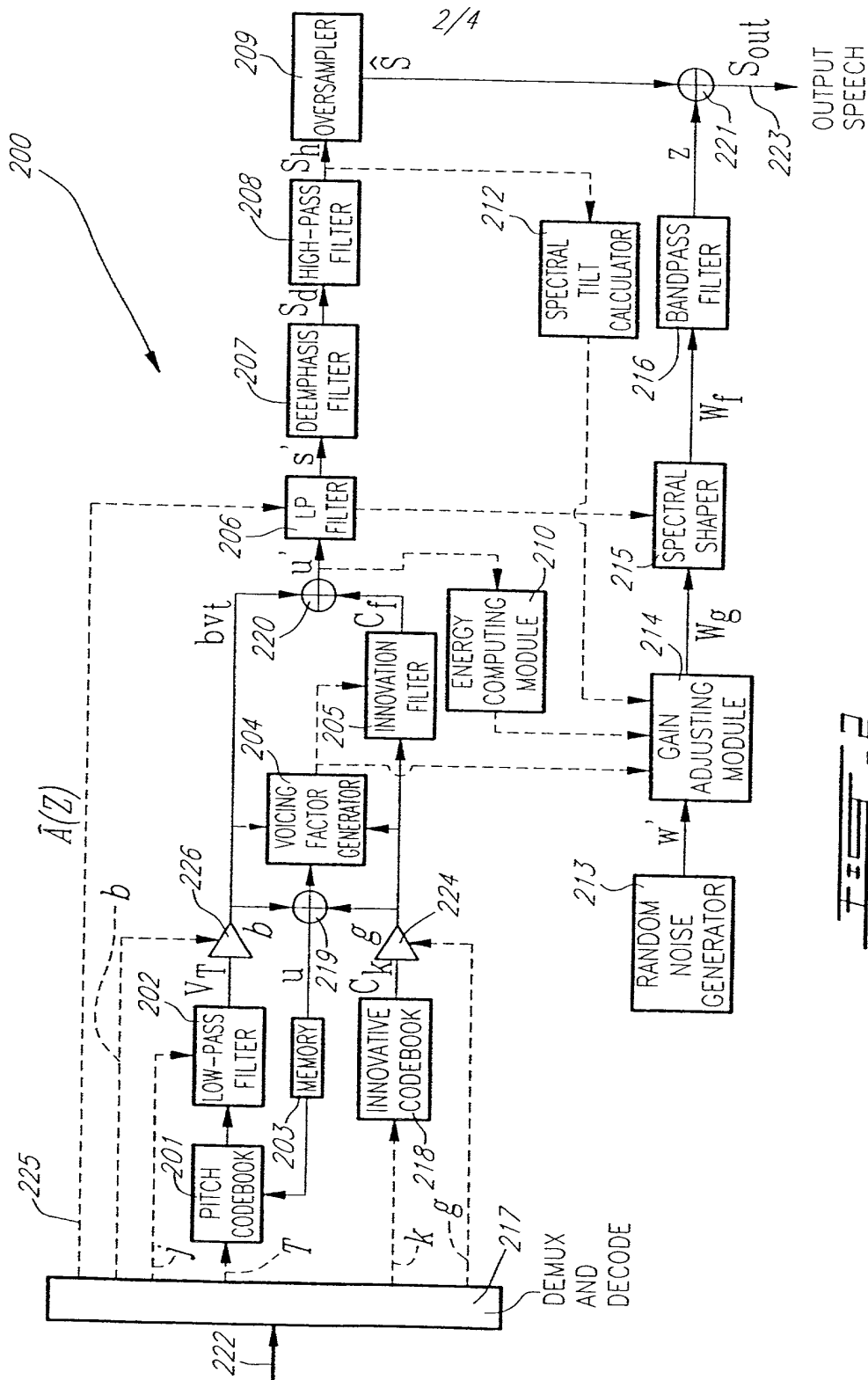
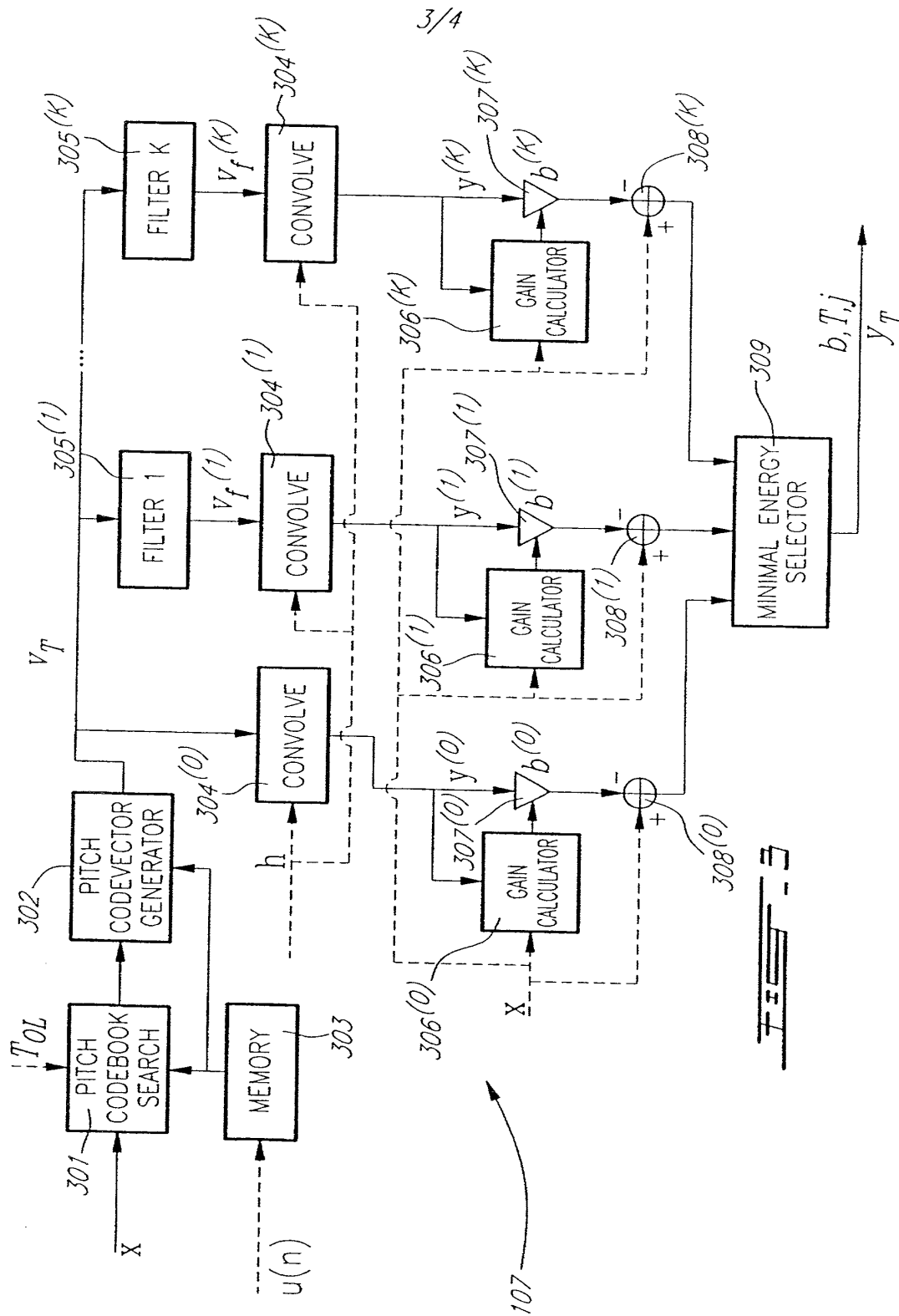
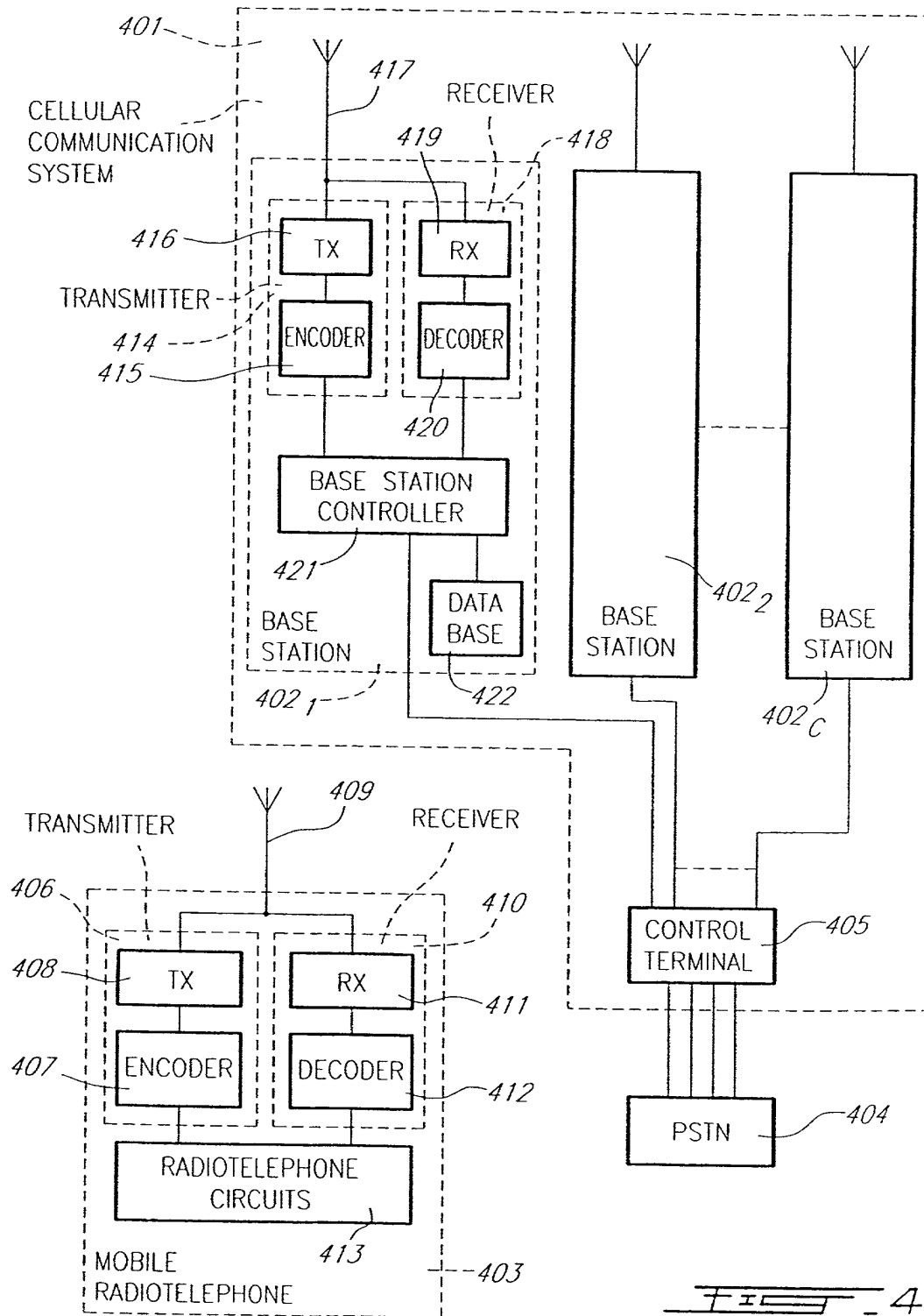


FIG. 2



4/4



BIRCH, STEWART, KOLASCH & BIRCH, LLPP.O. Box 747 • Falls Church, Virginia 22040-0747
Telephone: (703) 205-8000 • Facsimile: (703) 205-8050PLEASE NOTE:
YOU MUST
COMPLETE THE
FOLLOWING**COMBINED DECLARATION AND POWER OF ATTORNEY
FOR PATENT AND DESIGN APPLICATIONS**

As a below named inventor, I hereby declare that: my residence, post office address and citizenship are as stated next to my name; that I verily believe that I am the original, first and sole inventor (if only one inventor is named below) or an original, first and joint inventor (if plural inventors are named below) of the subject matter which is claimed and for which a patent is sought on the invention entitled:

Insert Title:

A METHOD AND DEVICE FOR ADAPTIVE BANDWIDTH PITCH SEARCH IN CODING WIDEBAND SIGNALSFill in Appropriate
Information -
For Use Without
Specification
Attached:

the specification of which is attached hereto. If not attached hereto,

the specification was filed on April 24, 2001 as

United States Application Number _____;

and amended on _____ (if applicable) and/or

the specification was filed on October 27, 1999 as PCTInternational Application Number PCT/CA99/01008; and was

amended under PCT Article 19 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 patentability as defined in Title 37, Code of Federal Regulations, §1.56.

I do not know and do not believe the same was ever known or used in the United States of America before my or our invention thereof, or patented or described in any printed publication in any country before my or our invention thereof or more than one year prior to this application, that the same was not in public use or on sale in the United States of America more than one year prior to this application, that the invention has not been patented or made the subject of an inventor's certificate issued before the date of this application in any country foreign to the United States of America on an application filed by me or my legal representative or assigns more than twelve months (six months for designs) prior to this application, and that no application for patent or inventor's certificate on this invention has been filed in any country foreign to the United States of America prior to this application by me or my legal representatives or assigns, except as follows.

I hereby claim foreign priority benefits under Title 35, United States Code, §119(a)-(d) 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:

Insert Priority
Information:
(if appropriate)**Prior Foreign Application(s)****Priority Claimed**

<u>2,252,170</u>	<u>Canada</u>	<u>27 October 1998</u>	☒	☐
(Number)	(Country)	(Month/Day/Year Filed)	Yes	No
<u> </u>	<u> </u>	<u> </u>	☐	☐
(Number)	(Country)	(Month/Day/Year Filed)	Yes	No
<u> </u>	<u> </u>	<u> </u>	☐	☐
(Number)	(Country)	(Month/Day/Year Filed)	Yes	No
<u> </u>	<u> </u>	<u> </u>	☐	☐
(Number)	(Country)	(Month/Day/Year Filed)	Yes	No

I hereby claim the benefit under Title 35, United States Code, §119(e) of any United States provisional applications(s) listed below.

Insert Provisional
Application(s):
(if any)

<u> </u>	<u> </u>
(Application Number)	(Filing Date)

<u> </u>	<u> </u>
(Application Number)	(Filing Date)

All Foreign Applications, if any, for any Patent or Inventor's Certificate Filed More than 12 Months (6 Months for Designs) Prior to the Filing Date of This Application:

Country	Application Number	Date of Filing (Month/Day/Year)
<u> </u>	<u> </u>	<u> </u>
<u> </u>	<u> </u>	<u> </u>

Insert Requested
Information:
(if appropriate)

I hereby claim the benefit under Title 35, United States Code, §120 of any United States and/or PCT 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 and/or PCT application in the manner provided by the first paragraph of Title 35, United States Code, §112, I acknowledge the duty to disclose information which is material to the patentability as defined in Title 37, Code of Federal Regulations, §1.56 which became available between the filing date of the prior application and the national or PCT international filing date of this application.

Insert Prior U.S.
Application(s):
(if any)

<u> </u>	<u> </u>	<u> </u>
(Application Number)	(Filing Date)	(Status - patented, pending, abandoned)

<u> </u>	<u> </u>	<u> </u>
(Application Number)	(Filing Date)	(Status - patented, pending, abandoned)

I hereby appoint the following attorneys to prosecute this application and/or an international application based on this application and to transact all business in the Patent and Trademark Office connected therewith and in connection with the resulting patent based on instructions received from the entity who first sent the application papers to the attorneys identified below, unless the inventor(s) or assignee provides said attorneys with a written notice to the contrary:

20	Raymond C. Stewart (Reg. No. 21,066)	Terrell C. Birch (Reg. No. 19,382)
	Joseph A. Kolasch (Reg. No. 22,463)	James M. Slattery (Reg. No. 28,380)
	Bernard L. Sweeney (Reg. No. 24,448)	Michael K. Mutter (Reg. No. 29,680)
	Charles Gorenstein (Reg. No. 29,771)	Gerald M. Murphy, Jr. (Reg. No. 28,977)
	Leonard R. Svensson (Reg. No. 30,330)	Terry L. Clark (Reg. No. 32,644)
	Andrew D. Meikle (Reg. No. 32,868)	Marc S. Weiner (Reg. No. 32,181)
	Joe McKinney Muncy (Reg. No. 32,334)	Donald J. Daley (Reg. No. 34,313)
	John W. Bailey (Reg. No. 32,881)	John A. Castellano (Reg. No. 35,094)
	Gary D. Yacura (Reg. No. 35,416)	Thomas S. Auchterlonie (Reg. No. 32,275)
	Mark J. Nuell (Reg. No. 36,623)	F. Prince Butler (Reg. No. 25,666)

Send Correspondence to:

BIRCH, STEWART, KOLASCH & BIRCH, LLP
P.O. Box 747 • Falls Church, Virginia 22040-0747
Telephone: (703) 205-8000 • Facsimile: (703) 205-8050

or Customer No. 2292

PLEASE NOTE:
YOU MUST
COMPLETE
THE
FOLLOWING:

Full Name of First or Sole Inventor:
Insert Name of Inventor:
Insert Date This Document is Signed:

Insert Residence:
Insert Citizenship:

Insert Post Office Address:

Full Name of Second Inventor, if any:
see above

Full Name of Third Inventor, if any:
see above

Full Name of Fourth Inventor, if any:
see above

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.

GIVEN NAME/FAMILY NAME		INVENTOR'S SIGNATURE	DATE*
Bruno BESSETTE		<i>Bruno Bessette</i>	June 6, 2001
Residence (City, State & Country)		CITIZENSHIP	
1546 Perodeau, Rock Forest, Quebec J1N 1L2, CANADA		CA	Canadian
MAILING ADDRESS (Complete Street Address including City, State & Country)			
Same as above			
GIVEN NAME/FAMILY NAME		INVENTOR'S SIGNATURE	DATE*
Redwan SALAMI		<i>Redwan Salami</i>	June 6, 2001
Residence (City, State & Country)		CITIZENSHIP	
963 Leo Laliberte, Sherbrooke, Quebec J1J 4L3, CANADA		CA	Canadian
MAILING ADDRESS (Complete Street Address including City, State & Country)			
Same as above			
GIVEN NAME/FAMILY NAME		INVENTOR'S SIGNATURE	DATE*
Roch LEFEBVRE		<i>Roch Lefebvre</i>	June 6, 2001
Residence (City, State & Country)		CITIZENSHIP	
259, avenue de la Bourgade, Canton de Magog, Quebec J1K 5R9, CANADA		CA	Canadian
MAILING ADDRESS (Complete Street Address including City, State & Country)			
Same as above			
GIVEN NAME/FAMILY NAME		INVENTOR'S SIGNATURE	DATE*
Residence (City, State & Country)		CITIZENSHIP	
MAILING ADDRESS (Complete Street Address including City, State & Country)			

*DATE OF SIGNATURE



JC17 PCT/PTO 20 JUN 2001 *PCT*
09/830-114

BOX PCT
PATENT
4082-0129P

IN THE U.S. PATENT AND TRADEMARK OFFICE

APPLICANT: Bruno BESSETTE et al.
INTERNATIONAL APPL. NO.: PCT/CA99/01008
APPL. NO.: 09/830,114
FILED: April 24, 2001;
Int'l: October 27, 1999
FOR: A METHOD AND DEVICE FOR
ADAPTIVE BANDWIDTH PITCH
SEARCH IN CODING WIDEBAND
SIGNALS

LETTER SUBMITTING ADDITIONAL DOCUMENTS FOR
ENTERING NATIONAL PHASE FOR A PCT APPLICATION

BOX PCT

Assistant Commissioner for Patents
Washington, DC 20231

June 20, 2001

Sir:

Under the provisions of 37 C.F.R. § 1.494 or 37 C.F.R. § 1.495, attached hereto are the following additional items necessary for entering the national phase in connection with the above-identified PCT international application.

- ☒ Executed Declaration and Power of Attorney.
- ☒ Original ☐ Photocopy
- ☐ The specification attached to the executed Declaration and Power of Attorney is a true copy of the specification which was filed in the U.S. Patent and Trademark Office on April 24, 2001, including any

Appl. No. 09/830,114

amendments thereto (if applicable) filed on even date therewith.

☒ The undersigned hereby declares that "Attorney Docket No. 4082-0129P" on page 1 of the attached inventors' Declaration corresponds to Appl. No. 09/830,114 filed April 24, 2001 entitled "A METHOD AND DEVICE FOR ADAPTIVE BANDWIDTH PITCH SEARCH IN CODING WIDEBAND SIGNALS."

☐ English language specification, claims, and Abstract with () sheets of drawings.

☒ Applicant claims small entity status under 37 C.F.R. § 1.27.

☐ Attached is a copy of Form PCT/DO/EO/905.

☐ _____

☐ No extension fee is required because the undersigned has not yet received the Notification of Missing Requirements (Form PCT/DO/EO/905). However, if for some reason it is determined that an extension of time is necessary, applicant hereby respectfully petitions for an extension of time for the filing of the present paper in accordance with the provisions of 37 C.F.R. § 1.136 and 37 C.F.R. § 1.17.

☐ Applicant(s) hereby respectfully petitions for () month(s) extension of time for the filing of the present paper in accordance with the provisions of 37 C.F.R. § 1.136 and 37 C.F.R. § 1.17. The required fee of \$0.00 is attached hereto.

Appl. No. 09/830,114

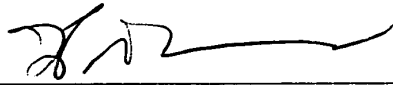
The Government Filing Surcharge in the amount of \$0.00 in accordance with 37 C.F.R. §§ 1.494 and 1.492 was previously paid for concurrently with the filing of the application on .

- ☒ Submitted concurrently herewith **under separate cover** for recording is an Assignment.
- ☐ A check in the amount of \$0.00 to cover the above-mentioned fees is enclosed.
- ☐ A Fee of \$0.00 to cover the increase in fees of the filing Surcharge is enclosed.

If necessary, the Commissioner is hereby authorized in this, concurrent, and future replies, to charge payment or credit any overpayment to Deposit Account No. 02-2448 for any additional fee required under 37 C.F.R. §§ 1.16 or 1.17; particularly, extension of time fees.

Respectfully submitted,

BIRCH, STEWART, KOLASCH & BIRCH, LLP

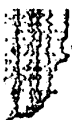
By 
F. Prince Butler, #25,666

FPB/tm
4082-0129P

Attachments

P.O. Box 747
Falls Church, VA 22040-0747
(703) 205-8000

(Rev. 01/22/01)



UNITED STATES PATENT AND TRADEMARK OFFICE

Commissioner for Patents, Box PC1
United States Patent and Trademark Office
Washington, D.C. 20231
www.uspto.gov

U.S. APPLICATION NO

09/830114

FIRST NAMED APPLICANT

BESSETTE

ATTY. DOCKET NO

4082-0129P

INTERNATIONAL APPLICATION NO

PCT/CA99/01008

I.A. FILING DATE

27 OCT 99

PRIORITY DATE

27 OCT 98

DATE MAILED

22 MAY 2001

BIRCH STEWART KOLASCH & BIRCH
PO BOX 747
FALLS CHURCH, VA 22040 0747**NOTIFICATION OF MISSING REQUIREMENTS UNDER 35 U.S.C. 371 IN THE UNITED STATES DESIGNATED/ELECTED OFFICE (DO/EO/US)**

1. The following items have been submitted by the applicant or the IB to the United States Patent and Trademark Office as ☐ a Designated Office (37 CFR 1.494) ☒ an Elected Office (37 CFR 1.495):

- ☒ U.S. Basic National Fee. ☒ Indication of Small Entity Status.
☒ Copy of the international application. ☐ Translation of the international application into English.
☐ Oath or Declaration of inventors(s). ☐ Translation of Article 19 amendments into English.
☐ Copy of Article 19 amendments. ☐ Other:
☐ Priority Document.
☐ The International Preliminary Examination Report in English and its Annexes, if any.
☐ Translation of Annexes to the International Preliminary Examination Report into English.

2. ☒ Applicant has requested early processing under 35 U.S.C. 371(f) but has not filed the following indicated items and/or the indicated items in paragraph 3 below. The Basic National Fee and the copy of the international application must be filed prior to 20 or 30 months from the priority date to avoid abandonment.

- ☐ U.S. Basic National Fee. ☐ Copy of the international application.

3. The following items **MUST** be furnished within the period set forth below in order to complete the requirements for acceptance under 35 U.S.C. 371:

- ☐ a. Translation of the application into English. A processing fee will be required if submitted later than the appropriate 20 or 30 months from the priority date.
- The current translation is defective for the reasons indicated on the attached Notice of Defective Translation.
☐ b. Processing fee for providing the translation of the application and/or the Annexes later than the appropriate 20 or 30 months from the priority date (37 CFR 1.492(f)).
☒ c. Oath or declaration of the inventors, in compliance with 37 CFR 1.497(a) and (b), properly identifying the application (preferably by the International application number and international filing date). A surcharge will be required if submitted later than the appropriate 20 or 30 months from the priority date.
- The current oath or declaration does not comply with 37 CFR 1.497(a) and (b) for the reasons indicated on the attached PCT/DO/EO/917.
☐ d. Surcharge for providing the oath or declaration later than the appropriate 20 or 30 months from the priority date (37 CFR 1.492(e)).

4. Additional claim fees of \$ _____ as a ☐ large entity ☐ small entity, including any required multiple dependent claim fee, are required. Applicant must submit the additional claim fees or cancel the additional claims for which fees are due (37 CFR 1.492(g)). See attached PTO-875.

5. ☐ Applicant has not submitted the required sequence listing pursuant to 37 CFR 1.821-1.825. See attached PCT/DO/EO/920.

ALL OF THE ITEMS SET FORTH IN 3(a)-3(d), 4 AND 5 ABOVE MUST BE SUBMITTED WITHIN TWO (2) MONTHS FROM THE DATE OF THIS NOTICE OR BY 22 OR 32 MONTHS (where 37 CFR 1.495 applies) FROM THE PRIORITY DATE FOR THE APPLICATION, WHICHEVER IS LATER. FAILURE TO PROPERLY RESPOND WILL RESULT IN ABANDONMENT.

The time period set above may be extended by filing a petition and fee for extension of time under the provisions of 37 CFR 1.136(a).

6. If box 3a or 3c is checked, a translation of the Annexes **MUST** be submitted no later than the time period set above or the Annexes will be cancelled. A processing fee will be required if submitted later than 20 or 30 months from the priority date.
7. ☐ The Article 19 amendments are cancelled since a translation was not provided by the appropriate 20 (37 CFR 1.494(d)) or 30 (37 CFR 1.495(d)) months from the priority date.

Applicant is reminded that any communication to the United States Patent and Trademark Office must be mailed to the address given in the heading and include the U.S. application no. shown above. (37 CFR 1.5)

A copy of this notice *MUST* be returned with this response.

Enclosed: ☐ PCT/DO/EO/917 ☐ Notice of Defective Translation
☐ PTO-875 ☐ PCT/DO/EO/920

Anita D. Johnson

FORM PCT/DO/EO/905 (March 2001)

Telephone: 703-305-3661

7-2201 ES
DOCKETED
Perfect



UNITED STATES PATENT AND TRADEMARK OFFICE

Commissioner for Patents, Box PCT
United States Patent and Trademark Office
Washington, D.C. 20231
www.uspto.gov

U.S. APPLICATION NO.	FIRST NAMED APPLICANT	ATTY. EXCISE NO.
09/830114	BESSETTE	B 4082-0129P
INTERNATIONAL APPLICATION NO.		
PCT/CA99/01008		
I.A. FILING DATE:	PRIORITY DATE:	
27 OCT 99	27 OCT 98	

BIRCH STEWART KOLASCH & BIRCH
PO BOX 747
FALLS CHURCH, VA 22040 0747

DATE MAILED: **29 JUN 2001**

**NOTIFICATION OF ACCEPTANCE OF APPLICATION UNDER 35 U.S.C. 371
AND 37 CFR 1.494 OR 1.495**

1. The applicant is hereby advised that the United States Patent and Trademark Office in its capacity as ☐ a Designated Office (37 CFR 1.494), ☒ an Elected Office (37 CFR 1.495), has determined that the above-identified international application has met the requirements of 35 U.S.C. 371, and is **ACCEPTED** for national patentability examination in the United States Patent and Trademark Office.

2. The United States Application Number assigned to the application is shown above and the relevant dates are:

20 June 2001	20 June 2001
DATE OF RECEIPT OF	DATE OF RECEIPT OF ALL
35 U.S.C. 371(c)(1), (c)(2) and (c)(4) REQUIREMENTS	35 U.S.C. 371 REQUIREMENTS

A Filing Receipt (PTO-103X) will be issued for the present application in due course. **THE DATE APPEARING ON THE FILING RECEIPT AS THE "FILING DATE" IS THE DATE ON WHICH THE LAST OF THE 35 U.S.C. 371 REQUIREMENTS HAS BEEN RECEIVED IN THE OFFICE. THIS DATE IS SHOWN ABOVE.** The filing date of the above-identified application is the international filing date of the international application (Article 11(3) and 35 U.S.C. 363). Once the Filing Receipt has been received, send all correspondence to the Group Art Unit designated thereon.

3. ☒ A request for immediate examination under 35 U.S.C. 371(f) was received on 24 April 2001 and the application will be examined in turn.

4. The following items have been received:

- ☒ U.S. Basic National Fee.
- ☒ Copy of the international application.
- ☐ Translation of the international application into English.
- ☒ Oath or Declaration of inventor(s).
- ☐ Copy of Article 19 amendments. ☐ Translation of Article 19 amendments into English.
The Article 19 amendments ☐ have ☐ not been entered.
- ☒ The International Preliminary Examination Report in English and its Annexes, if any.
- ☐ Copy of the Annexes to the International Preliminary Examination Report (IPER).
☐ Translation of Annexes to the IPER into English.
The Annexes ☐ have ☐ not been entered.
- ☒ Preliminary amendment(s) filed 24 April 2001 and _____.
- ☒ Information Disclosure Statement(s) filed 24 April 2001 and _____.
- ☐ Assignment document.
- ☒ Power of Attorney and/or Change of Address.
- ☐ Substitute specification filed _____.
- ☐ Indication of Small Entity Status.
- ☒ Priority Document.
- ☒ Copy of the International Search Report ☒ and copies of the references cited therein.
- ☐ Other:

Applicant is reminded that any communication to the United States Patent and Trademark Office must be mailed to the address given in the heading and include the U.S. application no. shown above (37 CFR 1.5).

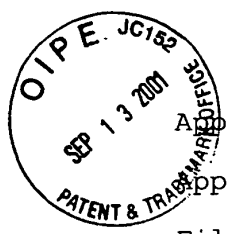
Deborah Williams *DW*

Telephone: 703-305-3744

FORM PCT/DO/EO/903 (March 2001)

#7
10/15/01
✓

PATENT
4082-0129PCT



IN THE U.S. PATENT AND TRADEMARK OFFICE

Applicant: Bruno BESSETTE et al. Conf.: 3008
Appl. No.: 09/830,114 Group: 2681
Filed: June 20, 2001 Examiner: UNKNOWN
For: METHOD AND DEVICE FOR ADAPTIVE
BANDWIDTH PITCH SEARCH IN CODING
WIDEBAND SIGNALS

RECEIVED
SEP 17 2001
Technology Center 2600

INFORMATION DISCLOSURE STATEMENT
(SUBMISSION AFTER FILING OF AN APPLICATION
BUT BEFORE FINAL REJECTION OR NOTICE OF ALLOWANCE
OR CONCURRENTLY WITH A RULE 53(d) CPA APPLICATION
OR WITH A RULE 1.114 RCE APPLICATION)

Assistant Commissioner for Patents
Washington, DC 20231

September 13, 2001

Sir:

Pursuant to 37 C.F.R. §§ 1.97 and 1.98, applicant(s) hereby submit(s) an Information Disclosure Statement for consideration by the Examiner.

I. LIST OF PATENTS, PUBLICATIONS OR OTHER INFORMATION

The patents, publications, or other information submitted for consideration by the Office are listed on the PTO-1449(s), attached hereto.

II. COPIES (check at least one box)

- a. ☒ Submitted herewith is a legible copy of (i) each U.S. and foreign patent; (ii) each publication or that portion which caused it to be listed; and (iii) all other information or that portion which caused it to be listed.
- b. ☐ Some or all of the documents listed on the PTO-1449 are not enclosed because they were cited in the International Search Report and copies should already be in the PTO file. If copies are needed, please contact the undersigned.

III. CONCISE EXPLANATION OF THE RELEVANCE
(check at least one box)

a. ☒ **DOCUMENTS IN THE ENGLISH LANGUAGE**

The attached patents, publications, or other information in the English language do not require a statement of relevancy.

b. ☐ **DOCUMENTS NOT IN THE ENGLISH LANGUAGE**

A concise explanation of the relevance of all patents, publications, or other information listed that is not in the English language is as follows:

c. ☐ **ENGLISH LANGUAGE SEARCH REPORT**

An English language version of the search report or action that indicates the degree of relevance found by the foreign office is attached, thereby satisfying the requirement for a concise explanation. See MPEP 609(A)(3).

d. ☒ **OTHER**

The following additional information is provided for the Examiner's consideration.

The article from IEEE Transaction ASSP is cited on page 19, line 12 of the instant specification; U.S. Patent Nos. 5,444,816; 5,699,482; 5,754,976 and 5,701,392 are all cited on page 30 of the present application.

FEEES

IV. ☒ THIS IDS IS BEING FILED UNDER 37 C.F.R. § 1.97(b):
(check one box)

- a. ☒ within three months of the filing date of a national application (37 C.F.R. § 1.97(b)(1)). No fee or statement is required. (This section is not to be used with RCE's and CPA's).
- b. ☐ within three months of the date of entry of the national stage as set forth in § 1.491 in an international application (37 C.F.R. § 1.97(b)(2)). No fee or statement is required.
- c. ☐ concurrently with the filing of a Continued Prosecution Application under 37 C.F.R. § 1.53(d) or concurrently with the filing of a Request for Continued Examination under § 1.114 (37 C.F.R. § 1.97(b)(4)). No fee or statement is required.
- d. ☐ before the mailing date of a first Action on the merits (37 C.F.R. § 1.97(b)(3)). No fee or statement is required.
In the event that a first Office Action on the merits has been issued, please consider this IDS under 37 C.F.R. § 1.97(c) and see the statement under 37 C.F.R. § 1.97(e) below, or, if no statement has been made, charge our deposit account in the amount of \$180.00 as required by 37 C.F.R. § 1.17(p).

V. ☐ THIS IDS IS BEING FILED UNDER 37 C.F.R. § 1.97(c):
(check one box)

before the mailing date of a Final Office Action under 37 C.F.R. § 1.113 (See 37 C.F.R. § 1.97(c)(1)) or before the mailing date of a Notice of Allowance under 37 C.F.R. § 1.311 (See 37 C.F.R. § 1.97(c)(2)).

- a. ☐ No statement; therefore, a fee in the amount of \$180.00 as required by 37 C.F.R. § 1.17(p).
- or
- b. ☐ See the statement below. No fee is required.

VI. STATEMENT UNDER 37 C.F.R. § 1.97(e) (check only one box)

The undersigned hereby states that

- a. ☐ each item of information contained in the IDS was first cited in any communication from a foreign Patent Office in a counterpart foreign application not more than three months prior to the filing of this IDS; or
- b. ☐ no item of information contained in the IDS was cited in a communication from a foreign Patent Office in a counterpart foreign application, and, to the knowledge of the person signing the certification after making reasonable inquiry, no item of IDS was known to any individual designated in 37 C.F.R. § 1.56(c) more than three months prior to the filing of the IDS.
- c. ☐ Some of the items of information were cited in a communication from a foreign Patent Office. As to this information, the undersigned states that each item of information contained in the IDS was cited in a communication from a foreign Patent Office in a counterpart foreign application not more than three months prior to the filing of this IDS. As to the remaining information, the undersigned hereby states that no item of this remaining information contained in the IDS was cited in a communication from a foreign Patent Office in a counterpart foreign application and, to the best of my knowledge after making reasonable inquiry, was known to any individual designated in 37 C.F.R. § 1.56(c) more than three months prior to the filing of this statement.

VII. PAYMENT OF FEES (check one box)

- ☐ A check in the amount of \$180.00 as required by 37 C.F.R. § 1.17(p) is enclosed for the above-identified fee.
- ☐ Please charge Deposit Account No. 02-2448 in the amount required by 37 C.F.R. § 1.17(p) for the above-indicated fee. A triplicate copy of this paper is attached.
- ☒ No fee is required.

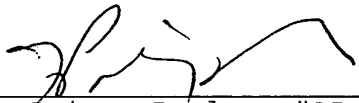
Appl. No. 09/830,114

If the Examiner has any questions concerning this IDS, he/she is requested to contact the undersigned. If it is determined that this IDS has been filed under the wrong rule, the PTO is requested to consider this IDS under the proper rule and charge the appropriate fee to Deposit Account No. 02-2448.

If necessary, the Commissioner is hereby authorized in this, concurrent, and future replies, to charge payment or credit any overpayment to Deposit Account No. 02-2448 for any additional fees required under 37 C.F.R. § 1.16 or under § 1.17; particularly, extension of time fees.

Respectfully submitted,

BIRCH, STEWART, KOLASCH & BIRCH, LLP

By 
F. Prince Butler, #25,666

FPB/glh
4082-0129PCT

P.O. Box 747
Falls Church, VA 22040-0747
(703) 205-8000

Enclosures: ☒ PTO-1449
☒ Documents
☐ Foreign Search Report
☐ Fee
☐ Other:

(Rev. 03/08/01)



#8 DAF \$ (S)
PATENT
4510-0105P

IN THE U.S. PATENT AND TRADEMARK OFFICE

Applicant: BASSETTE, et al. Conf.: 3008
Appl. No.: 09/830,114 Group: 2681
Filed: June 20, 2001
For: METHOD AND DEVICE FOR ADAPTIVE BANDWIDTH
PITCH SEARCH IN CODING WIDEBAND SIGNALS

RECEIVED
JUL 02 2003
OFFICE OF PETITIONS

NOTIFICATION OF ERRONEOUS PAYMENT OF SMALL ENTITY FEES
UNDER 37 C.F.R. § 1.28

Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

June 30, 2003

Adjustment date: 07/14/2003 AKELLEY
04/26/2001 UEDB1FE 00000045 09830114
01 FC:971 -430.00 OP
02 FC:254 -65.00 OP
03 FC:967 -387.00 OP

It is respectfully noted that the above-identified application does not qualify for small entity status under 37 C.F.R. § 1.27(a).

Small entity status was established with the filing of the application on June 20, 2001, and the small entity filing fee was paid in good faith. It has since been discovered that this status as a small entity was established in error, without the intent to deceive.

It is therefore respectfully requested that the error be excused. The following items are being submitted to comply with the requirements of 37 C.F.R. 1.28(c):

- (1) Separate submission for each application or patent.

This request is being submitted in connection with Patent Appl. No. 09/830,114.

- (2) Payment of deficiency owed.

Adjustment date: 07/14/2003 AKELLEY
07/02/2003 AMONDAF1 00000019 09830114
01 FC:1461 -922.00 OP

A \$922.00 small entity filing fee and additional claims fee under 37 C.F.R. §

07/02/2003 AMONDAF1 00000019 09830114

01 FC:1461

922.00 OP

09830114
136.00 OP
774.00 OP
900.00 OP
07/14/2003 AKELLEY
01 FC:1617
02 FC:1615
03 FC:1613

Appl. No. 60/366,243

1.482 and \$65.00 Surcharge Fee under § 1.492(e) was paid on June 20, 2001, with check number 4017451. Accordingly, the deficiency owed is \$922.00.

- (ii) A check in the amount of \$922.00 is attached hereto, as payment of the deficiency owed.

It is again noted that the previous establishment of small entity status was made in error, without the intent to deceive. As it is believed that the requirements of 37 C.F.R. § 1.28(c)(1) and (c)(2) have been met, it is respectfully requested that the error be excused and that the U.S. Patent and Trademark Office forward appropriate notification to the undersigned.

If necessary, the Commissioner is hereby authorized in this, concurrent, and future replies, to charge payment or credit any overpayment to Deposit Account No. 02-2448 for any additional fees required under 37 C.F.R. § 1.16 or under 37 C.F.R. § 1.17; particularly, extension of time fees.

Respectfully submitted,

BIRCH, STEWART, KOLASCH & BIRCH, LLP

By 

Marc S. Weiner, #32,181

MSW/HNS/ndb
4510-0106P

P.O. Box 747
Falls Church, VA 22040-0747
(703) 205-8000



UNITED STATES PATENT AND TRADEMARK OFFICE

COMMISSIONER FOR PATENTS
UNITED STATES PATENT AND TRADEMARK OFFICE
P.O. Box 1450
ALEXANDRIA, VA 22313-1450
www.uspto.gov

Paper No. 9

BIRCH STEWART KOLASCH & BIRCH
P.O. BOX 747
FALLS CHURCH, VA 22040-0747

COPY MAILED

In re Application of : JUL 11 2003
Bruno Bessette et al :
Application No. 09/830,114 : NOTICE OFFICE OF PETITIONS
Filed: June 20, 2001 :
Attorney Docket No. 4082-0129P :

This is a notice regarding your request for acceptance of a fee deficiency submission under 37 CFR 1.28. On September 1, 1998, the Court of Appeals for the Federal Circuit held that 37 CFR 1.28(c) is the sole provision governing the time for correction of the erroneous payment of the issue fee as a small entity. See DH Technology v. Synergystex International, Inc. 154 F.3d 1333, 47 USPQ2d 1865 (Fed. Cir. Sept. 1, 1998).

The Office no longer investigates or rejects original or reissue applications under 37 CFR 1.56. 1098 Off. Gaz. Pat. Office 502 (January 3, 1989). Therefore, nothing in this Notice is intended to imply that an investigation was done.

Your fee deficiency submission under 37 CFR 1.28 is hereby **ACCEPTED.**

Inquiries related to this communication should be directed to the Office of Petitions Staff at (703) 305-9285.

This file is being forwarded to Technology Center AU 2682.

Wan Laymon
Petitions Examiner
Office of Petitions
Office of the Deputy Commissioner
for Patent Examination Policy



PATENT
4510-0105P

IN THE U.S. PATENT AND TRADEMARK OFFICE

Applicant: BESSETTE, Bruno et al. Conf.: 3008
Appl. No.: 09/830,114 Group: 2642
Filed: June 20, 2001 Examiner: AHMAD MATAR
For: METHOD AND DEVICE FOR ADAPTIVE
BANDWIDTH PITCH SEARCH IN CODING
WIDEBAND SIGNALS

RECEIVED

AUG 04 2004

INFORMATION DISCLOSURE STATEMENT
(SUBMISSION AFTER FILING OF AN APPLICATION
BUT BEFORE FINAL REJECTION OR NOTICE OF ALLOWANCE
OR CONCURRENTLY WITH A RULE 1.114 RCE APPLICATION)

Technology Center 2600

Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

August 3, 2004

Sir:

Pursuant to 37 C.F.R. §§ 1.97 and 1.98, applicant(s) hereby submit(s) an Information Disclosure Statement for consideration by the Examiner.

I. LIST OF PATENTS, PUBLICATIONS OR OTHER INFORMATION

The patents, publications, or other information submitted for consideration by the Office are listed on the PTO-1449(s), attached hereto.

II. COPIES (check at least one box)

- a. ☒ This application was filed before June 30, 2003. Accordingly, submitted herewith is a legible copy of (i) each U.S. and foreign patent; (ii) each publication or that portion which caused it to be listed; and (iii) all other information or that portion which caused it to be listed.
- b. ☐ This application was filed on or after June 30, 2003. Accordingly, copies of cited U.S. patents and patent application publications therefore are not included. Copies of foreign

Best Available Copy

patent documents and non-patent literature are included.

- c. ☐ Some or all of the documents listed on the PTO-1449 are not enclosed because they were cited in the International Search Report and copies should already be in the PTO file. If copies are needed, please contact the undersigned.

III. CONCISE EXPLANATION OF THE RELEVANCE
(check at least one box)

- a. ☐ **DOCUMENTS IN THE ENGLISH LANGUAGE**

The patents, publications, or other information listed on the attached PTO 1449 are in the English language and therefore, do not require a statement of relevancy.

- b. ☒ **DOCUMENTS NOT IN THE ENGLISH LANGUAGE**

A concise explanation of the relevance of all patents, publications, or other information listed that is not in the English language is as follows:

The foreign reference has an English translation attached.

- c. ☐ **ENGLISH LANGUAGE SEARCH REPORT**

An English language version of the search report or action that indicates the degree of relevance found by the foreign office is attached, thereby satisfying the requirement for a concise explanation. See MPEP 609(III)(A)(3).

- d. ☐ **OTHER**

The following additional information is provided for the Examiner's consideration.

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FÉES

- IV. ☒ THIS IDS IS BEING FILED UNDER 37 C.F.R. § 1.97(b):
(check one box)
- a. ☐ within three months of the filing date of a national application (37 C.F.R. § 1.97(b)(1)). No fee or statement is required. *(This section is not to be used with RCE's.)*
 - b. ☐ within three months of the date of entry of the national stage as set forth in § 1.491 in an international application (37 C.F.R. § 1.97(b)(2)). No fee or statement is required.
 - c. ☐ concurrently with the filing of a Request for Continued Examination under § 1.114 (37 C.F.R. § 1.97(b)(4)). No fee or statement is required.
 - d. ☒ before the mailing date of a first Action on the merits (37 C.F.R. § 1.97(b)(3)). No fee or statement is required.
In the event that a first Office Action on the merits has been issued, please consider this IDS under 37 C.F.R. § 1.97(c) and see the statement under 37 C.F.R. § 1.97(e) below, or, if no statement has been made, charge our deposit account in the amount of \$180.00 as required by 37 C.F.R. § 1.17(p).
- V. ☐ THIS IDS IS BEING FILED UNDER 37 C.F.R. § 1.97(c):
(check one box)
- before the mailing date of a Final Office Action under 37 C.F.R. § 1.113 (See 37 C.F.R. § 1.97(c)(1)) or before the mailing date of a Notice of Allowance under 37 C.F.R. § 1.311 (See 37 C.F.R. § 1.97(c)(2)).
- a. ☐ No statement; therefore, a fee in the amount of \$180.00 as required by 37 C.F.R. § 1.17(p).
 - or
 - b. ☐ See the statement below. No fee is required.

³ Best Available Copy

VI. STATEMENT UNDER 37 C.F.R. § 1.97(e) (check only one box)

The undersigned hereby states that

- a. ☐ each item of information contained in the IDS was first cited in any communication from a foreign Patent Office in a counterpart foreign application not more than 30 days prior to the filing of this IDS; or
- b. ☐ each item of information contained in the IDS was first cited in any communication from a foreign Patent Office in a counterpart foreign application not more than three months prior to the filing of this IDS; or
- c. ☐ no item of information contained in the IDS was cited in a communication from a foreign Patent Office in a counterpart foreign application, and, to the knowledge of the person signing the certification after making reasonable inquiry, no item of IDS was known to any individual designated in 37 C.F.R. § 1.56(c) more than three months prior to the filing of the IDS.
- d. ☐ Some of the items of information were cited in a communication from a foreign Patent Office. As to this information, the undersigned states that each item of information contained in the IDS was first cited in a communication from a foreign Patent Office in a counterpart foreign application not more than three months prior to the filing of this IDS. As to the remaining information, the undersigned hereby states that no item of this remaining information contained in the IDS was cited in a communication from a foreign Patent Office in a counterpart foreign application and, to the best of my knowledge after making reasonable inquiry, was known to any individual designated in 37 C.F.R. § 1.56(c) more than three months prior to the filing of this statement.

4 Best Available Copy

VII. PAYMENT OF FEES (check one box)

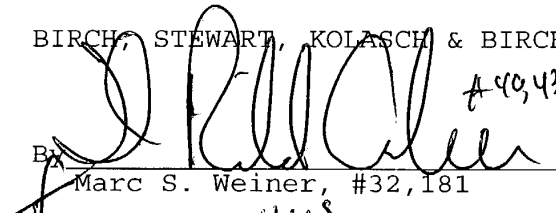
- ☐ A check in the amount of \$180.00 as required by 37 C.F.R. § 1.17(p) is enclosed for the above-identified fee.
- ☐ Please charge Deposit Account No. 02-2448 in the amount required by 37 C.F.R. § 1.17(p) for the above-indicated fee. A triplicate copy of this paper is attached.
- ☒ No fee is required.

If the Examiner has any questions concerning this IDS, he/she is requested to contact the undersigned. If it is determined that this IDS has been filed under the wrong rule, the PTO is requested to consider this IDS under the proper rule and charge the appropriate fee to Deposit Account No. 02-2448.

If necessary, the Commissioner is hereby authorized in this, concurrent, and future replies, to charge payment or credit any overpayment to Deposit Account No. 02-2448 for any additional fees required under 37 C.F.R. § 1.16 or under § 1.17; particularly, extension of time fees.

Respectfully submitted,

BIRCH, STEWART, KOLASCH & BIRCH, LLP


By Marc S. Weiner, #32,181

P.O. Box 747
Falls Church, VA 22040-0747
(703) 205-8000

MSW/HNS/fjl
4510-0105P

Attachment(s): ☒ PTO-1449
☒ Documents
☐ Foreign Search Report
☐ Fee
☐ Other:

(Rev. 05/14/2004)

AUG 03 2004

Sheet 1 of 1 (8/3/04)

Form PTO-1449

INFORMATION DISCLOSURE CITATION
IN AN APPLICATION

(Use several sheets if necessary)

ATTY. DOCKET NO.
4510-0105PAPPLICATION NO.
09/830,114APPLICANT
BESSETTE, Bruno et al.FILING DATE
June 20, 2001GROUP
2642

U.S. PATENT DOCUMENTS

EXAMINER INITIAL	DOCUMENT NUMBER	Kind	DATE	NAME	CLASS	SUB CLASS	FILING DATE IF APPROPRIATE
	US						
	US						
	US						
	US						
	US						
	US						
	US						
	US						
	US						

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AUG 04 2004

Technology Center 2600

FOREIGN PATENT DOCUMENTS

Office	DOCUMENT NUMBER	Kind	DATE	COUNTRY	CLASS	SUB CLASS	TRANSLATION	
							YES	NO

OTHER DOCUMENTS

(Include Name of the author (in CAPITAL LETTERS), title of the article (when appropriate), title of the item (book, magazine, journal, serial, symposium, catalog, etc.) date, page(s), volume-issue number(s), publisher, city and/or country where published.

M.V. NAZAROV et al., "Techniques of Digital Processing and Transmission of Voice Signals"; Moscow, Radio i Svyaz' publishers 1985, pp. 10-12.

EXAMINER

DATE CONSIDERED

EXAMINER: Initial if citation considered, whether or not citation is in conformance with M.P.E.P. 609; Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

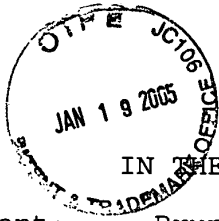
MSW/HNS/fjl

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PLUS Search Results for S/N 09830114, Searched November 30, 2004

The Patent Linguistics Utility System (PLUS) is a USPTO automated search system for U.S. Patents from 1971 to the present. PLUS is a query-by-example search system which produces a list of patents that are most closely related linguistically to the application searched. This search was prepared by the staff of the Scientific and Technical Information Center, SIRA.

6795805	4268720	6108369	4535473
6807524	4311877	6128374	4331837
6615169	4401854	6222875	4410764
6691085	4449231	6222875	6205190
6732070	4471445	6246885	6219342
5646961	4797950	6289308	6219342
5699382	4835517	6310704	4053712
5235669	4918736	6366604	4330689
5321793	4935963	6373927	4479226
5710863	4943855	6389000	4815023
5778335	5019788	6393392	4864561
5787390	5068899	6591100	4924492
5790759	5200955	6606312	4955037
5873060	5201292	6611798	5001703
5937378	5303346	6671514	5274711
6014621	5353376	6700958	5438565
6134518	5459729	6704711	5455888
6141638	5459730	6778649	5659157
6182030	5473613	6792267	5737485
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5924062	5515396	6415252	5859840
6662154	5546420	5602961	5872775
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6055496	5581652	5659659	5903554
5875423	5592517	5884253	6005855
6249758	5604927	5913187	6035209
5903866	5619524	6052659	6064663
5924061	5627833	6113653	6137840
6138092	5668837	6188979	6175590
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6351490	RE35740	6226607	6675144
6424940	5729613	6233550	6691083
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6801887	5751739	6324503	5933803
5530655	5752225	6324505	5724410
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5222189	5844597	6397178	5664054
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5777992	5970154	6475245	5893060
5963899	5978759	6584438	5909663
6173162	6006108	6640209	5987406
3639691	6034971	6678649	6073092
3875341	6044073	6754630	5195168
4131765	6047022	6810381	5233660



2654
JFW

PATENT
4510-0105P

IN THE U.S. PATENT AND TRADEMARK OFFICE

Applicant: Bruno BESSETTE et al. Conf.: 3008
Appl. No.: 09/830,114 Group: 2654
Filed: June 20, 2001 Examiner: A. ARMSTRONG
For: A METHOD AND DEVICE FOR ADAPTIVE
BANDWIDTH PITCH SEARCH IN CODING
WIDEBAND SIGNALS

STATUS INQUIRY

Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

JAN 19 2005

Sir:

Please advise the undersigned as to the status of the above-identified application as follows:

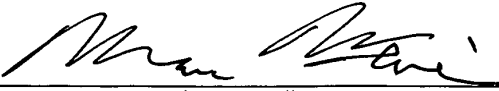
Status of Office Action

A duplicate copy of the present Status Inquiry is attached hereto. It is respectfully requested that the duplicate copy be marked appropriately to indicate the status of the above-identified application and returned to the undersigned as soon as possible.

Please call the undersigned if there are any questions with regard to the present request.

Respectfully submitted,

BIRCH, STEWART, KOLASCH & BIRCH, LLP

By 
Marc S. Weiner, #32,181

MSW/HNS/bsh
4510-0105P

P.O. Box 747
Falls Church, VA 22040-0747
(703) 205-8000

- ☐ Awaiting Action from Patent Office Examiner - Expected Date for Action is: _____.
- ☐ Awaiting Decision by Board of Appeals.
- ☐ Application allowed on _____.
- ☐ Abandoned per _____.
- ☐ Other _____.

Ref #	Hits	Search Query	DBs	Default Operator	Plurals	Time Stamp
L1	291	pitch with predict\$ with error	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/06/27 14:19
L2	43	convol\$5 with pitch with \$4vector	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/06/27 14:19
L3	968	weight\$ with synthesi\$ with filter	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/06/27 14:20
L4	8	2 same 3	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/06/27 14:20
L5	2	"5621852".pn.	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/06/27 15:12
S2	197	("6795805" "6807524" "6615169" "6691085" "6732070" "5646961" "5699382" "5235669" "5321793" "5710863" "5778335" "5787390" "5790759" "5873060" "5937378" "6014621" "6134518" "6141638" "6182030" "5845244" "5717825" "5826226" "5924062" "6662154" "5890110" "6055496" "5875423" "6249758" "5903866" "5924061" "6138092" "6192335" "6351490" "6424940" "6631347" "6801887" "5530655" "5539829" "5606618" "5806025" "5222189" "5357594" "5388992" "5550924" "5777992" "5963899" "6173162" "3639691" "3875341" "4131765").pn. ("4268720" "4311877" "4401854" "4449231" "4471445" "4797950" "4835517" "4918736" "4935963" "4943855" "5019788" "5068899" "5200955" "5201292" "5303346" "5353376" "5459729" "5459730" "5473613" "5479562" "5483600" "5487174" "5515396" "5546420" "5577117" "5581652" "5592517" "5604927" "5619524" "5627833" "5668837" "5706335" RE35740 "5729613" "5745856" "5751739" "5752225" "5787119" "5799013" "5802169" "5844597" "5878038" "5886276" "5970058" "5970154" "5978759" "6006108" "6034971" "6044073" "6047022").pn. ("6108369" "6128374" "6222875" "6222875" "6246885" "6289308" "6310704" "6366604" "6373927" "6389000" "6393392" "6591100" "6606312" "6611798" "6671514" "6700958" "6704711" "6778649" "6792267" "5537410" "5673266" "6141353" "6415252" "5602961" "5729655" "5659659" "5884253" "5913187" "6052659" "6113653" "6188979" "6226607" "6226607" "6233550" "6260017" "6324503" "6324505" "6330532" "6393394" "6397175" "6397178" "6434519" "6438518" "6449592" "6475245" "6584438" "6640209" "6678649" "6754630" "6810381").pn. ("4535473" "4331837" "4410764" "6205190" "6219342" "6219342" "4053712" "4330689" "4479226" "4815023" "4864561" "4924492" "4955037" "5001703" "5274711" "5438565" "5455888" "5659157" "5737485" "5751725" "5777990" "5850394" "5859840" "5872775" "5901145" "5903554" "6005855" "6035209" "6064663" "6137840" "6175590" "6385190" "6675144" "6691083" "6782367" "5933803" "5724410" "5909662" "5528723" "5633982" "5664054" "5706392" "5719993" "5781881" "5893060" "5909663" "5987406" "6073092" "5195168" "5233660").pn.	US-PGPUB; USPAT	OR	ON	2005/05/25 19:09
S3	194	pitch with prediction with error	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/05/25 19:14
S4	331	pitch same codebook same search	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/05/25 19:17

Search History 6/27/05 4:27:58 PM Page 1

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S5	54	S3 and S4	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/05/25 19:17
S6	14	("4932061" "5307441" "5359696" "5444816" "5519807" "5664055" "5699482" "5701392" "5754976" "5963898" "6006174" "6064962" "6192334" "6449590").PN. OR ("6807524").URPN.	US-PGPUB; USPAT; USOCR	OR	ON	2005/05/25 19:48
S7	150	filter\$5 same codevector	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2005/06/23 18:26
S8	0	("2005/0108007").URPN.	USPAT	OR	ON	2005/06/23 19:12
S9	14	("4932061" "5307441" "5359696" "5444816" "5519807" "5664055" "5699482" "5701392" "5754976" "5963898" "6006174" "6064962" "6192334" "6449590").PN. OR ("6807524").URPN.	US-PGPUB; USPAT; USOCR	OR	ON	2005/06/23 19:18
S10	7	("5235669" "5444816" "5450449" "5699482" "5701392" "5754976" "5819213").PN.	US-PGPUB; USPAT; USOCR	OR	ON	2005/06/23 19:23



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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/830,114	06/20/2001	Bruno Bessette	4082-0129P	3008
2292	7590	07/01/2005	EXAMINER	
BIRCH STEWART KOLASCH & BIRCH PO BOX 747 FALLS CHURCH, VA 22040-0747			ARMSTRONG, ANGELA A	
			ART UNIT	PAPER NUMBER
			2654	

DATE MAILED: 07/01/2005

Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No.	Applicant(s)	
	09/830,114	BESSETTE ET AL.	
	Examiner	Art Unit	
	Angela A. Armstrong	2654	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1) ☒ Responsive to communication(s) filed on 20 June 2001.

2a) ☐ This action is **FINAL**. 2b) ☒ This action is non-final.

3) ☐ 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.

Disposition of Claims

4) ☒ Claim(s) 1-63 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5) ☐ Claim(s) _____ is/are allowed.

6) ☒ Claim(s) 1-4, 7, 9-13, 16, 18-22, 25, 27-31, 34, 36-40, 43, 45-49, 52, 54-58, 61 and 63 is/are rejected.

7) ☒ Claim(s) 5, 6, 8, 14, 15, 17, 23, 24, 26, 32, 33, 35, 41, 42, 44, 50, 51, 53, 59, 60 and 62 is/are objected to.

8) ☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9) ☐ The specification is objected to by the Examiner.

10) ☐ The drawing(s) filed on _____ is/are: a) ☐ accepted or b) ☐ objected to by the Examiner.
Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).
Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11) ☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12) ☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).
a) ☐ All b) ☐ Some * c) ☐ None of:
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)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08)
Paper No(s)/Mail Date 4/24/01; 9/13/01; 8/03/04

4) ☐ Interview Summary (PTO-413)
Paper No(s)/Mail Date. _____

5) ☐ Notice of Informal Patent Application (PTO-152)

6) ☐ Other: _____

DETAILED ACTION

Claim Rejections - 35 USC § 102

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.

1. Claims 1, 3, 7, 10, 12, 16, 19, 21, and 25 are rejected under 35 U.S.C. 102(b) as being anticipated by Lin (US Patent No. 5,621,852).
2. Regarding claims 1 and 10, Lin discloses an efficient codebook structure for code excited linear prediction coding and provides support for a pitch analysis device for producing an optimal set of pitch codebook parameters, comprising: a) at Least two signal paths associated to respective sets of pitch codebook parameters (binary codebook 1 and binary codebook 2), wherein: i) each signal path comprises a pitch prediction error calculating device for calculating a pitch predictor error of a pitch codevector from a pitch codebook search device(Figure 3 and 4); and ii) at Least one of said two paths comprises a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of said one path (50,68, 74, 84); and b) a selector for comparing the pitch prediction errors calculated in said at Least two signal paths, for choosing the signal path having the Lowest calculated pitch prediction error, and for selecting the set of pitch codebook parameters associated to the chosen signal path (Figure 3, 4).

Regarding claims 3 and 12, Lin discloses signal paths comprises a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same path (50,68, 74, 84).

Regarding claims 7 and 16, Lin discloses pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the Lowest calculated energy of the pitch prediction error (Figure 3, 4).

Regarding claim 19, Lin discloses an encoder (Figure 1) having a pitch analysis device as in claim 1 (figure 3, 4) for encoding a wideband input signal, said encoder comprising: a) a Linear prediction synthesis filter (26) calculator responsive to the wideband signal for producing Linear prediction synthesis filter coefficients; b) a perceptual weighting filter (12), responsive to the wideband signal and the Linear prediction synthesis filter coefficients, for producing a perceptually weighted signal', c) an impulse response generator responsive to said Linear prediction synthesis filter coefficients for producing a weighted synthesis filter impulse response signal (16, 22); d) a pitch search unit for producing pitch codebook parameters, said pitch search unit comprising: i) said pitch codebook search device responsive to the perceptually weighted signal and the linear prediction synthesis filter coefficients for producing the pitch codevector and an innovative search target vector (18); and ii) said pitch analysis device responsive to the pitch codevector for selecting, from said sets of pitch codebook parameters, the set of pitch codebook parameters associated to the path having the Lowest calculated pitch prediction error

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(18,24); d) an innovative codebook search device, responsive to the weighted synthesis filter impulse response signal, and the innovative search target vector, for producing innovative codebook parameters (18,24); and e) a signal forming device for producing an encoded wideband signal comprising the set of pitch codebook parameters associated to the path having the Lowest pitch prediction error, said innovative codebook parameters, and said Linear prediction synthesis filter coefficients (30).

Regarding claim 21, Lin discloses signal paths comprises a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same path (50,68, 74, 84).

Regarding claim 25, Lin discloses pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the Lowest calculated energy of the pitch prediction error (Figure 3, 4).

Claim Rejections - 35 USC § 103

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 negated by the manner in which the invention was made.

3. Claims 2, 4, 9, 11, 13, 18, 20, 22, 27, 28-31, 34, 36, 37-40, 43, 45, 46-49, 52, 54, 55-58, 61, and 63 are rejected under 35 U.S.C. 103(a) as being unpatentable over Lin in view of Adoul (US Patent No. 5,754,976).

4. Regarding claim 28, Lin does not teach the specifics of an environment for using the CELP encoding technique to include a cellular communication system. However, implementing a CELP coding technique in the cellular communication environment was well known in the art, as evidenced by Adoul. Adoul teaches implementation of the CELP encoder in a cellular communication system for servicing a large geographical area divided into a plurality of cells, comprising, mobile portable transmitter/receiver units, cellular base stations respectively situated in the cells, means for controlling communication between the cellular base station, a bidirectional wireless communication sub-system between each mobile unit (col. 3, lines 31-48). Adoul specifically teaches the advantages of using a CELP encoder in the wireless communication/telecommunications platform so as to realize the good quality/bit rate tradeoff associated with the CELP coder. It would have been obvious to one of ordinary skill at the time of the invention to modify the system of Lin to provide for implementation in a wireless platform, as suggested by Adoul, so as to provide the improved encoding of Lin in a wireless environment.

Regarding claim 29, Lin discloses, one of the paths comprises no filter for filtering the pitch codevector (figure 5, 103).

Regarding claims 30, 31, and 36 Lin discloses Lin discloses signal paths comprises a plurality of signal paths each provided with a filter for filtering the pitch codevector before

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supplying said pitch codevector to the pitch prediction error calculating device of the same path (50,68, 74, 84).

Regarding claim 34, Lin discloses pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the Lowest calculated energy of the pitch prediction error (Figure 3, 4).

Regarding claims 2, 4, 9, 11, 13, 18, 20, 22, 27, 37-40, 43, 45, 46-49, 52, 54, 55-58, 61, and 63, claims 2, 4, 9, 11, 13, 18, 20, 22, 27, 37-40, 43, 45, 46-49, 52, 54, 55-58, 61, and 63 are similar in scope and content to claims 1, 3, 7, 28-31 and 34-36, and are therefore rejected under similar rationale.

Allowable Subject Matter

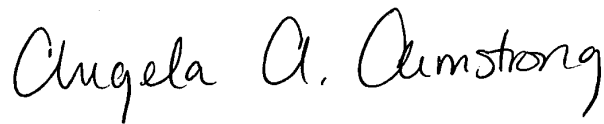
5. Claims 5-6, 8, 14-15, 17, 23-24, 26, 32-33, 35, 41-42, 44, 50-51, 53, 59-60, and 62 objected to as being dependent upon a rejected base claim, but would be allowable if rewritten in independent form including all of the limitations of the base claim and any intervening claims.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Angela A. Armstrong whose telephone number is 571-272-7598. The examiner can normally be reached on Monday-Thursday 11:30-8:00 PM.

Art Unit: 2654

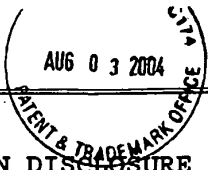
If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on 571-272-7602. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see <http://pair-direct.uspto.gov>. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).



Angela A Armstrong
Examiner
Art Unit 2654

AAA
June 27, 2005



Form PTO-1449	ATTY. DOCKET NO. 4510-0105P	APPLICATION NO. 09/830,114
INFORMATION DISCLOSURE CITATION IN AN APPLICATION (Use several sheets if necessary)	APPLICANT BESSETTE, Bruno et al.	
	FILING DATE June 20, 2001	GROUP 2642

U.S. PATENT DOCUMENTS

EXAMINER INITIAL	DOCUMENT NUMBER	Kind	DATE	NAME	CLASS	SUB CLASS	FILING DATE IF APPROPRIATE
	US						
	US						
	US						
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FOREIGN PATENT DOCUMENTS

	Office	DOCUMENT NUMBER	Kind	DATE	COUNTRY	CLASS	SUB CLASS	TRANSLATION	
								YES	NO

OTHER DOCUMENTS (Include Name of the author (in CAPITAL LETTERS), title of the article (when appropriate), title of the item (book, magazine, journal, serial, symposium, catalog, etc.) date, page(s), volume-issue number(s), publisher, city and/or country where published.

AAA	M.V. NAZAROV et al., "Techniques of Digital Processing and Transmission of Voice Signals"; Moscow, Radio i Svyaz' publishers 1985, pp. 10-12.

EXAMINER <i>Angela Armstrong</i>	DATE CONSIDERED <i>6/23/05</i>
EXAMINER: Initial if citation considered, whether or not citation is in conformance with M.P.E.P. 609; Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.	

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Notice of References Cited	Application/Control No. 09/830,114	Applicant(s)/Patent Under Reexamination BESSETTE ET AL.	
	Examiner Angela A. Armstrong	Art Unit 2654	Page 1 of 1

U.S. PATENT DOCUMENTS

*		Document Number Country Code-Number-Kind Code	Date MM-YYYY	Name	Classification
	A	US-5,621,852	04-1997	Lin, Daniel	704/219
	B	US-5,754,976	05-1998	Adoul et al.	704/223
	C	US-			
	D	US-			
	E	US-			
	F	US-			
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	M	US-			

FOREIGN PATENT DOCUMENTS

*		Document Number Country Code-Number-Kind Code	Date MM-YYYY	Country	Name	Classification
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NON-PATENT DOCUMENTS

*		Include as applicable: Author, Title Date, Publisher, Edition or Volume, Pertinent Pages)
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	X	

*A copy of this reference is not being furnished with this Office action. (See MPEP § 707.05(a).)
Dates in MM-YYYY format are publication dates. Classifications may be US or foreign.

Index of Claims 	Application/Control No.		Applicant(s)/Patent under Reexamination	
	09/830,114		BESSETTE ET AL.	
	Examiner		Art Unit	
	Angela A. Armstrong		2654	

✓	Rejected
=	Allowed

—	(Through numeral) Cancelled
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N	Non-Elected
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Claim		Date											
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Search Notes

Application/Control No.

09/830,114

Examiner

Angela A. Armstrong

Applicant(s)/Patent under
Reexamination

BESSETTE ET AL.

Art Unit

2654

SEARCHED

Class	Subclass	Date	Examiner
704	219,220, 223	5/25/2005	AAA

INTERFERENCE SEARCHED

Class	Subclass	Date	Examiner

**SEARCH NOTES
(INCLUDING SEARCH STRATEGY)**

	DATE	EXMR
plus search	5/25/2005	AAA
east text search	5/25/2005	AAA
consultation with T. Smits	6/23/2005	AAA
consultation with V. Chawan	6/23/2005	AAA
east text search	6/23/2005	AAA
east text search	6/27/2005	AAA



Attorney Docket: 4510-0105P

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicants: Bruno BESSETTE et al. Conf.: 3008
Appl No: 09/830,114 Art Unit: 2654
Filed: June 20, 2001 Examiner: A. Armstrong
For: METHOD AND DEVICE FOR ADAPTIVE BANDWIDTH PITCH
SEARCH IN CODING WIDEBAND SIGNALS

REPLY UNDER 37 C.F.R. § 1.111

Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

November 1, 2005

Sir:

In reply to the Examiner's Non-Final Office Action dated July 1, 2005, the period for reply having been extended one (1) month, the following amendments and remarks are respectfully submitted in connection with the above-identified application as follows:

- Amendments to the claims;
- Remarks/Arguments.

AMENDMENTS TO THE CLAIMS

Please amend the claims as follows:

1. (CURRENTLY AMENDED) A pitch analysis device for producing an optimal a set of pitch codebook parameters, comprising:

a) at least two signal paths associated to respective sets of pitch codebook parameters, wherein:

i) each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device; and

ii) at least one of said two signal paths comprises a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of said one signal path; and

b) a selector for comparing the pitch prediction errors calculated in said at least two signal paths, for choosing the signal path having the lowest calculated pitch prediction error and for selecting the set of pitch codebook parameters associated to the chosen signal path.

2. (CURRENTLY AMENDED) A pitch analysis device as defined in claim 1, wherein one of said at least two signal paths comprises no filter for

filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

3. (CURRENTLY AMENDED) A pitch analysis device as defined in claim 1, wherein said signal paths comprise a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same signal path.

4. (CURRENTLY AMENDED) A pitch analysis device as defined in claim 3, wherein the filters of said plurality of signal paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

5. (ORIGINAL) A pitch analysis device as defined in claim 1, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with a weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

6. (ORIGINAL) A pitch analysis device as defined in claim 5, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x' y^{(j)} / \|y^{(j)}\|^2$$

where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

7. (ORIGINAL) A pitch analysis device as defined in claim 1, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said

pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

8. (ORIGINAL) A pitch analysis device as defined in claim 5, wherein:
- a) each of said filters of the plurality of signal paths is identified by a filter index;
 - b) said pitch codevector is identified by a pitch codebook index; and
 - c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

9. (ORIGINAL) A pitch analysis device as defined in claim 1, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

10. (CURRENTLY AMENDED) A pitch analysis method for producing ~~an optimal~~ a set of pitch codebook parameters, comprising:

a) in at least two signal paths associated to respective sets of pitch codebook parameters, calculating, for each signal path, a pitch prediction error of a pitch codevector from a pitch codebook search device;

b) in at least one of said two signal paths, filtering the pitch codevector before supplying said pitch codevector for calculation of said pitch prediction error of said one signal path; and

c) comparing the pitch prediction errors calculated in said at least two signal paths, choosing the signal path having the lowest calculated pitch prediction error, and selecting the set of pitch codebook parameters associated to the chosen signal path.

11. (CURRENTLY AMENDED) A pitch analysis method as defined in claim 10, wherein, in one of said at least two signal paths, no filtering of the pitch codevector is performed before supplying said pitch codevector to a pitch prediction error calculating device.

12. (CURRENTLY AMENDED) A pitch analysis method as defined in claim 10, wherein said signal paths ~~comprises~~ comprise a plurality of signal

paths and wherein filtering the pitch codevector is performed in each of said plurality of signal paths before supplying said pitch codevector to the pitch prediction error calculating device of the same signal path.

13. (CURRENTLY AMENDED) A pitch analysis method as defined in claim 12, further comprising selecting the filters of said plurality of signal paths from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

14. (ORIGINAL) A pitch analysis method as defined in claim 10, wherein calculating a pitch prediction error in each signal path comprises:

- a) convolving the pitch codevector with a weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;
- b) calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;
- c) multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and
- d) combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

15. (ORIGINAL) A pitch analysis method as defined in claim 14, wherein said pitch gain calculation comprises calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$$

where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

16. (ORIGINAL) A pitch analysis method as defined in claim 10, wherein calculating said pitch prediction error, in each signal path, comprises calculating an energy of the corresponding pitch prediction error, and wherein comparing the pitch prediction ~~error~~ errors comprises comparing the energies of said pitch prediction errors of the different signal paths and choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

17. (ORIGINAL) A pitch analysis method as defined in claim 14, wherein:

a) identifying each of said filters of the plurality of signal paths by a filter index;

- b) identifying said pitch codevector by a pitch codebook index; and
- c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

18. (ORIGINAL) A pitch analysis method as defined in claim 10, wherein filtering of the pitch codevector is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

19. (CURRENTLY AMENDED) An encoder having a pitch analysis device as in claim 1 for encoding a wideband input signal, said encoder comprising:

- a) a linear prediction synthesis filter calculator responsive to the wideband signal for producing linear prediction synthesis filter coefficients;
- b) a perceptual weighting filter, responsive to the wideband signal and the linear prediction synthesis filter coefficients, for producing a perceptually weighted signal;
- c) an impulse response generator responsive to said linear prediction synthesis filter coefficients for producing a weighted synthesis filter impulse response signal;

d) a pitch search unit for producing pitch codebook parameters, said pitch search unit comprising:

i) said pitch codebook search device responsive to the perceptually weighted signal and the linear prediction synthesis filter coefficients for producing the pitch codevector and an innovative search target vector; and

ii) said pitch analysis device responsive to the pitch codevector for selecting, from said sets of pitch codebook parameters, the set of pitch codebook parameters associated to the signal path having the lowest calculated pitch prediction error;

e) an innovative codebook search device, responsive to a weighted synthesis filter impulse response signal, and the innovative search target vector, for producing innovative codebook parameters; and

f) a signal forming device for producing an encoded wideband signal comprising the set of pitch codebook parameters associated to the signal path having the lowest pitch prediction error, said innovative codebook parameters, and said linear prediction synthesis filter coefficients.

20. (CURRENTLY AMENDED) An encoder as defined in claim 19, wherein one of said at least two signal paths comprises no filter for filtering the

pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

21. (CURRENTLY AMENDED) An encoder as defined in claim 19, wherein said signal paths ~~comprises~~ comprise a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same signal path.

22. (CURRENTLY AMENDED) An encoder as defined in claim 21, wherein the filters of said plurality of signal paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

23. (ORIGINAL) An encoder as defined in claim 19, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

24. (ORIGINAL) An encoder as defined in claim 23, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$$

where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

25. (ORIGINAL) An encoder as defined in claim 19, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch

prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

26. (ORIGINAL) An encoder as defined in claim 23, wherein:

- a) each of said filters of the plurality of signal paths is identified by a filter index;
- b) said pitch codevector is identified by a pitch codebook index; and
- c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

27. (ORIGINAL) An encoder as defined in claim 19, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

28. (CURRENTLY AMENDED) A cellular communication system for servicing a large geographical area divided into a plurality of cells, comprising:

- a) mobile transmitter/receiver units;
- b) cellular base stations respectively situated in said cells;

c) a control terminal for controlling communication between the cellular base stations; and

d) a bidirectional wireless communication sub-system between each mobile unit situated in one cell and the cellular base station of said one cell, said bidirectional wireless communication sub-system comprising, in both the mobile unit and the cellular base station:

i) a transmitter including an encoder for encoding a wideband signal as recited in claim 19, and a transmission circuit for transmitting the encoded wideband signal; and

ii) a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal.

29. (CURRENTLY AMENDED) A cellular communication system as defined in claim 28, wherein one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

30. (CURRENTLY AMENDED) A cellular communication system as defined in claim 28, wherein said signal paths ~~comprises~~ comprise a plurality

of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same signal path.

31. (CURRENTLY AMENDED) A cellular communication system as defined in claim 30, wherein the filters of said plurality of signal paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

32. (ORIGINAL) A cellular communication system as defined in claim 28, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and the pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

33. (ORIGINAL) A cellular communication system as defined in claim 32, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$$

where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

34. (ORIGINAL) A cellular communication system as defined in claim 28, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

35. (ORIGINAL) A cellular communication system as defined in claim 32, wherein:

- a) each of said filters of the plurality of signal paths is identified by a filter index;
- b) said pitch codevector is identified by a pitch codebook index; and
- c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

36. (ORIGINAL) A cellular communication system as defined in claim 28, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

37. (ORIGINAL) A cellular mobile transmitter/receiver unit, comprising:

- a) a transmitter including an encoder for encoding a wideband signal as recited in claim 19 and a transmission circuit for transmitting the encoded wideband signal; and

b) a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal.

38. (CURRENTLY AMENDED) A cellular mobile transmitter/receiver unit as defined in claim 37, wherein one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

39. (CURRENTLY AMENDED) A cellular mobile transmitter/receiver unit as defined in claim 37, wherein said signal paths ~~comprises~~comprise a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same signal path.

40. (CURRENTLY AMENDED) A cellular mobile transmitter/receiver unit as defined in claim 39, wherein the filters of said plurality of signal paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

41. (ORIGINAL) A cellular mobile transmitter/receiver unit as defined in claim 37, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

42. (ORIGINAL) A cellular mobile transmitter/receiver unit as defined in claim 41, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$$

where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

43. (ORIGINAL) A cellular mobile transmitter/receiver unit as defined in claim 37, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

44. (ORIGINAL) A cellular mobile transmitter/receiver unit as defined in claim 41, wherein:

- a) each of said filters of the plurality of signal paths is identified by a filter index;
- b) said pitch codevector is identified by a pitch codebook index; and
- c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

45. (ORIGINAL) A cellular mobile transmitter/receiver unit as defined in claim 37, wherein said filter is integrated in an interpolation filter of said

pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

46. (CURRENTLY AMENDED) A ~~cellular~~ network element, comprising:

a) a transmitter including an encoder for encoding a wideband signal as recited in claim 19 and a transmission circuit for transmitting the encoded wideband signal ;and

~~b) a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal.~~

47. (CURRENTLY AMENDED) A ~~cellular~~ network element as defined in claim 46, wherein one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

48. (CURRENTLY AMENDED) A ~~cellular~~ network element as defined in claim 46, wherein said signal paths ~~comprises~~ comprise a plurality of signal paths each provided with a filter for filtering the pitch codevector before

supplying said pitch codevector to the pitch prediction error calculating device of the same ~~signal~~ path.

49. (CURRENTLY AMENDED) A ~~cellular~~ network element as defined in claim 48, wherein the filters of said plurality of signal paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

50. (CURRENTLY AMENDED) A ~~cellular~~ network element as defined in claim 46, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

51. (CURRENTLY AMENDED) A ~~cellular~~ network element as defined in claim 50, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$$

where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

52. (CURRENTLY AMENDED) A ~~cellular~~ network element as defined in claim 46, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

53. (CURRENTLY AMENDED) A ~~cellular~~ network element as defined in claim 50, wherein:

- a) each of said filters of the plurality of signal paths is identified by a filter index;
- b) said pitch codevector is identified by a pitch codebook index; and
- c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

54. (CURRENTLY AMENDED) A ~~cellular~~ network element as defined in claim 46, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

55. (CURRENTLY AMENDED) In a cellular communication system for servicing a ~~large~~ geographical area divided into a plurality of cells, comprising: mobile transmitter/receiver units, cellular base stations ~~—~~ respectively situated in said cells; and a control terminal for controlling communication between the cellular base stations;

a bidirectional wireless communication sub-system between each mobile unit situated in one cell and the cellular base station of said one cell, said

bidirectional wireless communication sub-system comprising, in both the mobile unit and the cellular base station:

a) a transmitter including an encoder for encoding a wideband signal as recited in claim 19, and a transmission circuit for transmitting the encoded wideband signal; and

b) a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal.

56. (CURRENTLY AMENDED) A bidirectional wireless communication sub-system as defined in claim 55, wherein one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

57. (CURRENTLY AMENDED) A bidirectional wireless communication sub-system as defined in claim 55, wherein said signal paths ~~comprises~~ comprise a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same signal path.

58. (CURRENTLY AMENDED) A bidirectional wireless communication sub-system as defined in claim 57, wherein the filters of said plurality of signal paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

59. (ORIGINAL) A bidirectional wireless communication sub-system as defined in claim 55, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

60. (ORIGINAL) A bidirectional wireless communication sub-system as defined in claim 59, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$$

where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

61. (ORIGINAL) A bidirectional wireless communication sub-system as defined in claim 55, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

62. (ORIGINAL) A bidirectional wireless communication sub-system as defined in claim 59, wherein:

a) each of said filters of the plurality of signal paths is identified by a filter index;

b) said pitch codevector is identified by a pitch codebook index; and

c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

63. (ORIGINAL) A bidirectional wireless communication sub-system as defined in claim 55, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

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REMARKS

Favorable reconsideration and allowance of the present application are respectfully requested in view of the following remarks. Claims 1-63 remain pending. Claims 1 and 10 are independent.

ALLOWABLE SUBJECT MATTER

Applicants appreciate that claims 5-6, 8, 14-15, 17, 23-24, 26, 32-33, 35, 41-42, 44, 50-51, 53, 59-60 and 62 are indicated to define allowable subject matter. *See Office Action, page 6, item 5.*

SCOPE OF CLAIMS NOT ALTERED

The claims have been amended merely to address informal issues and to enhance clarity. It is intended that the claim scope is not narrowed by the amendments.

§ 102 REJECTION – LIN

Claims 1, 3, 7, 10, 12, 16, 19, 21 and 25 stand rejected under 35 U.S.C. § 102(b) as allegedly being anticipated by Lin (US Patent 5,621,852). *See Office Action, pages 2-4, item 2.* Applicants respectfully traverse.

Independent claim 1 recites as follows:

1. A pitch analysis device for producing a set of pitch codebook parameters, comprising:

a) at least two signal paths associated to respective sets of pitch codebook parameters, wherein:

i) each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device; and

ii) at least one of said two signal paths comprises a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of said one signal path; and

b) a selector for comparing the pitch prediction errors calculated in said at least two signal paths, for choosing the signal path having the lowest calculated pitch prediction error and for selecting the set of pitch codebook parameters associated to the chosen signal path.

As will be demonstrated below, Lin cannot be relied upon to anticipate claim 1 as the Examiner alleges.

Lin describes a CELP encoder which represents the input speech samples as digital parameters comprising an LSP index, a pitch lag and gain, and a code index and gain, for digital multiplexing by transmitter 30 on communication channel 31 (column 4, lines 12-16). Lin also indicates that pitch lag and gain are derived from the input speech using a weighted

synthesis filter 16, and an adaptive codebook analysis 18 (column 4, lines 25-27).

Lin further indicates that:

- The innovation code index and gain is made adaptive to the voice of the speaker using a second weighted synthesis filter 22, and a ternary codebook analysis 24, containing an encoder ternary codebook (column 4, lines 48-51);
- The ternary codebook analysis 24 of Figure 1 is illustrated in further detail by the process flow diagram of Figure 3 (column 5, lines 32-33);
- Figure 3 illustrates an exhaustive search process for the optimum innovation sequence. All combinations of binary codevectors in binary codebooks 1 and 2 are computed for the fidelity criterion function $\Psi(i,j)$. The peak fidelity criterion function $\Psi(i,j)$ is selected at step 62, thereby identifying the desired codebook index i and codebook index j (column 6, lines 6-11); and
- As illustrated in Figure 6A-6B of Lin, the binary codevector from codebook 1 is added to the binary output from codebook 2 to form a ternary codevector constituting the innovation sequence.

In the Office Action of July 1st, 2005, the Examiner alleges that, regarding claims 1 and 10, Lin discloses an efficient codebook structure for

code excited linear prediction coding and provides support for a pitch analysis device for producing a set of pitch codebook parameters, comprising at least two signal paths associated to respective sets of pitch codebook parameters (binary codebook 1 and binary codebook 2).

As indicated hereinabove, Lin's binary codebooks 1 and 2 are not used to produce a set of pitch codebook parameters, but are used to search an optimum innovation sequence. The optimum innovation sequence is identified by indices i and j of the codebooks 1 and 2. Accordingly, the binary codebooks of Lin do not form a pitch analysis device nor do they form signal paths that are associated to respective sets of pitch codebook parameters.

It is well known that in the art of CELP coding, the pitch lag and gain and the innovation sequence are two different sets of coding parameters searched through different search processes. Since claim 1 relates to the production of a set of pitch codebook parameters, the subject matter of this claim cannot be anticipated by a technique for searching an optimum innovation sequence in a CELP encoder as taught by Lin. This alone is sufficient to distinguish claim 1 over Lin.

But in addition, Lin's binary codebooks 1 and 2 produce respective, partial contributions of a single innovation sequence. This is different from the

invention as claimed in claim 1 where each signal path is associated with a set of pitch codebook parameters.

Further, in the Office Action of July 1st, 2005, the Examiner alleges that Lin's Figures 3 and 4 teach signal paths each comprising a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector.

However, as indicated hereinabove, Figure 3 of Lin illustrates an exhaustive search process for the optimum innovation sequence (codebook index i and codebook index j). More specifically, Lin describes that the partial contributions of innovation sequence from the codebooks 1 and 2 are respectively supplied to correlation steps 52 and 72 in which correlations with the target vector are calculated. Therefore, prediction errors related to the innovation sequence are calculated. Accordingly and contrary to the Examiner's allegation, Lin describes no pitch prediction error calculating device in the signal paths associated to codebooks 1 and 2 as featured in claim 1.

In the same manner, Lin's filters 50 and 68 of Figure 3 processes partial contributions of an innovation sequence. Therefore, Lin describes no filter for filtering a pitch codevector in the signal paths as featured in claim 1.

Yet further, the peak selection 62 of Figure 3 of Lin is responsive to correlations between the innovation sequence contributions from codebooks 1

and 2 and the target vector. Accordingly, Lin describes no selector for comparing the pitch prediction errors calculated in the signal paths and for choosing the signal path having the lowest calculated pitch prediction error as featured in claim 1.

For at least the reasons stated above, independent claim 1 is distinguishable over Lin.

Like claim 1, claim 10 also relates to the production of a set of pitch codebook parameters. It has been amply demonstrated above that the subject matter of this claim cannot be anticipated by a technique for searching an optimum innovation sequence in a CELP encoder as taught by Lin. This alone is sufficient to distinguish claim 10 over Lin.

But in addition, claim 10 recites, in part “in at least two signal paths associated to respective sets of pitch codebook parameters,” “calculating, for each signal path, a pitch prediction error of a pitch codevector from a pitch codebook search device,” “in at least one of said two signal paths, filtering the pitch codevector” and “selecting the set of pitch codebook parameters associated to the chosen signal path.” It has been demonstrated above that Lin cannot teach or suggest any of these recited features. Therefore, independent claim 10 is also distinguishable over Lin.

Claims 3, 7, 12, 16, 19, 21 and 25 depend from independent claims 1 and 10 directly or indirectly. Therefore, for at least the reasons stated with respect to claims 1 and 10 as well as on their own merits, claims 3, 7, 12, 16, 19, 21 and 25 are also distinguishable over Lin.

Applicants respectfully requests that the rejection of claims 1, 3, 7, 10, 12, 16, 19, 21 and 25 based on Lin be withdrawn.

§ 103 REJECTION – PRIMARY, SECONDARY, TERTIARY

Claims 2, 4, 9, 11, 13, 18, 20, 22, 27, 28-31, 34, 36, 37-40, 43, 45, 46-49, 52, 54, 55-58, 61 and 63 stand rejected under 35 U.S.C. § 103(a) as allegedly being unpatentable over Lin in view of Adoul et al. (US Patent 5,754,976). *See Office Action, pages 5-6, items 3-4.* Applicants respectfully traverse.

The rejected claims depend from independent claims 1 and 10 directly or indirectly. It has been demonstrated above that claims 1 and 10 are distinguishable over Lin. Adoul has not been relied upon to correct for at least the above noted deficiencies of Lin. Therefore, independent claims 1 and 10 are distinguishable over the combination of Lin and Adoul.

For at least due to the dependency thereon from claims 1 and 10 as well as on their own merits, claims 2, 4, 9, 11, 13, 18, 20, 22, 27, 28-31, 34, 36,

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37-40, 43, 45, 46-49, 52, 54, 55-58, 61 and 63 are also distinguishable over the combination of Lin and Adoul.

Applicants respectfully request that the rejection of claims 2, 4, 9, 11, 13, 18, 20, 22, 27, 28-31, 34, 36, 37-40, 43, 45, 46-49, 52, 54, 55-58, 61 and 63 based on Lin and Adoul be withdrawn.

CONCLUSION

All objections and rejections raised in the Office Action having been addressed, it is respectfully submitted that the present application is in condition for allowance. Should there be any outstanding matters that need to be resolved, the Examiner is respectfully requested to contact Hyung Sohn (Reg. No. 44,346), to conduct an interview in an effort to expedite prosecution in connection with the present application.

Pursuant to 37 C.F.R. §§ 1.17 and 1.136(a), Applicants respectfully petition for a one (1) month extension of time for filing a reply in connection with the present application, and the required fee is attached hereto.

If necessary, the Commissioner is hereby authorized in this, concurrent, and future replies, to charge payment or credit any overpayment to Deposit

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Account No. 02-2448 for any additional fees required under 37 C.F.R. §§ 1.16
or 1.17; particularly, extension of time fees.

Respectfully submitted,

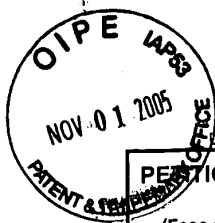
BIRCH, STEWART, KOLASCH &, BIRCH, LLP

By: 

Marc S. Weiner
Reg. No. 32,181

HNS
MSW/HNS/ags
4510-0105P

P.O. Box 747
Falls Church, VA 22040-0747
(703) 205-8000

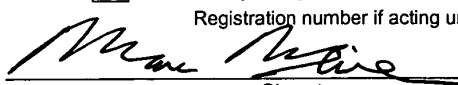


PTO/SB/22 (12-04)

Approved for use through 7/31/2006. OMB 0651-0031

U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE

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.

PECTION FOR EXTENSION OF TIME UNDER 37 CFR 1.136(a) FY 2005 (Fees pursuant to the Consolidated Appropriations Act, 2005 (H.R. 4818).)		Docket Number (Optional) 4510-0105P	
Application Number 09/830,114-Conf. #003008		Filed June 20, 2001	
For A METHOD AND DEVICE FOR ADAPTIVE BANDWIDTH PITCH SEARCH IN CODING WIDEBAND SIGNALS			
Art Unit 2654		Examiner A. A. Armstrong	
This is a request under the provisions of 37 CFR 1.136(a) to extend the period for filing a reply in the above identified application. The requested extension and fee are as follows (check time period desired and enter the appropriate fee below):			
		<u>Fee</u>	<u>Small Entity Fee</u>
☒	One month (37 CFR 1.17(a)(1))	\$120	\$60
☐	Two months (37 CFR 1.17(a)(2))	\$450	\$225
☐	Three months (37 CFR 1.17(a)(3))	\$1020	\$510
☐	Four months (37 CFR 1.17(a)(4))	\$1590	\$795
☐	Five months (37 CFR 1.17(a)(5))	\$2160	\$1080
☐	Applicant claims small entity status. See 37 CFR 1.27.	11/02/2005 SZEWDIE1 00000041 09830114	
☒	A check in the amount of the fee is enclosed.	01 FC:1251	120.00 OP
☐	Payment by credit card. Form PTO-2038 is attached.		
☐	The Director has already been authorized to charge fees in this application to a Deposit Account.		
☒	The Director is hereby authorized to charge any fees which may be required, or credit any overpayment, to Deposit Account Number 02-2448 . I have enclosed a duplicate copy of this sheet.		
I am the	☐ applicant/inventor.		
	☐ assignee of record of the entire interest. See 37 CFR 3.71. Statement under 37 CFR 3.73(b) is enclosed. (Form PTO/SB/96).		
	☐ attorney or agent of record. Registration Number _____		
	☒ attorney or agent under 37 CFR 1.34.		
	Registration number if acting under 37 CFR 1.34	32,181	
	 Signature	November 1, 2005	
	Marc S. Weiner	(703) 205-8000	
	Typed or printed name	Telephone Number	
NOTE: Signatures of all the inventors or assignees of record of the entire interest or their representative(s) are required. Submit multiple forms if more than one signature is required, see below.			
☐	Total of 1 forms are submitted.		



FEE TRANSMITTAL For FY 2005		Complete if Known	
Effective on 12/08/2004. Fees pursuant to the Consolidated Appropriations Act, 2005 (H.R. 4818).		Application Number	09/830,114-Conf. #003008
☐ Applicant claims small entity status. See 37 CFR 1.27		Filing Date	June 20, 2001
		First Named Inventor	Bruno BESSETTE
		Examiner Name	A. A. Armstrong
		Art Unit	2654
		Attorney Docket No.	4510-0105P
TOTAL AMOUNT OF PAYMENT			
(\$)		120.00	

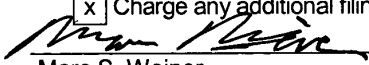
METHOD OF PAYMENT (check all that apply)	
☒ Check	☐ Credit Card
☐ Money Order	☐ None
☐ Other (please identify): _____	
☐ Deposit Account	Deposit Account Number: <u>02-2448</u> Deposit Account Name: <u>Birch, Stewart, Kolasch & Birch, LLP</u>
For the above-identified deposit account, the Director is hereby authorized to: (check all that apply)	
☐ Charge fee(s) indicated below	☐ Charge fee(s) indicated below, except for the filing fee
☒ Charge any additional fee(s) or underpayment of fee(s) under 37 CFR 1.16 and 1.17	☒ Credit any overpayments

FEE CALCULATION							
1. BASIC FILING, SEARCH, AND EXAMINATION FEES							
	FILING FEES		SEARCH FEES		EXAMINATION FEES		
		Small Entity		Small Entity		Small Entity	
Application Type	Fee (\$)	Fee (\$)	Fee (\$)	Fee (\$)	Fee (\$)	Fee (\$)	Fees Paid (\$)
Utility	300	150	500	250	200	100	
Design	200	100	100	50	130	65	
Plant	200	100	300	150	160	80	
Reissue	300	150	500	250	600	300	
Provisional	200	100	0	0	0	0	
2. EXCESS CLAIM FEES							
						Small Entity	
Fee Description						Fee (\$)	Fee (\$)
Each claim over 20 (including Reissues)						50	25
Each independent claim over 3 (including Reissues)						200	100
Multiple dependent claims						360	180
Total Claims		Extra Claims	Fee (\$)	Fee Paid (\$)	Multiple Dependent Claims		
63		- 63 =	x	=	Fee (\$)		Fee Paid (\$)
Indep. Claims		Extra Claims	Fee (\$)	Fee Paid (\$)			
2		- 3 =	x	=			
3. APPLICATION SIZE FEE							
If the specification and drawings exceed 100 sheets of paper (excluding electronically filed sequence or computer listings under 37 CFR 1.52(e)), the application size fee due is \$250 (\$125 for small entity) for each additional 50 sheets or fraction thereof. See 35 U.S.C. 41(a)(1)(G) and 37 CFR 1.16(s).							
Total Sheets		Extra Sheets	Number of each additional 50 or fraction thereof		Fee (\$)	Fee Paid (\$)	
- 100 =		/50	(round up to a whole number) x		=		
4. OTHER FEE(S)							
Non-English Specification, \$130 fee (no small entity discount)						Fees Paid (\$)	
Other (e.g., late filing surcharge): 1251 Extension for response within first month						120.00	

SUBMITTED BY			
Signature		Registration No. (Attorney/Agent)	32,181
Name (Print/Type)	Marc S. Weiner	Telephone	(703) 205-8000
		Date	November 1, 2005



2654
Dm

AMENDMENT TRANSMITTAL LETTER				Docket No. 4510-0105P	
Application No. 09/830,114-Conf. #003008		Filing Date June 20, 2001		Examiner A. A. Armstrong	
				Art Unit 2654	
Applicant(s): Bruno BESSETTE et al.					
Invention: A METHOD AND DEVICE FOR ADAPTIVE BANDWIDTH PITCH SEARCH IN CODING WIDEBAND SIGNALS					
MS Amendment Commissioner for Patents P.O. Box 1450 Alexandria, VA 22313-1450					
Transmitted herewith is an amendment in the above-identified application.					
The fee has been calculated and is transmitted as shown below.					
CLAIMS AS AMENDED					
	Claims Remaining After Amendment	Highest Number Previously Paid	Number Extra Claims Present	Rate	
Total Claims	63	- 63 =		x	
Independent Claims	2	- 3 =		x	
Multiple Dependent Claims (check if applicable) ☐					
Other fee (please specify): Extension for response within first month					120.00
TOTAL ADDITIONAL FEE FOR THIS AMENDMENT:					120.00
☒ Large Entity ☐ Small Entity					
☐ No additional fee is required for this amendment.					
☐ Please charge Deposit Account No. _____ in the amount of \$ _____. A duplicate copy of this sheet is enclosed.					
☒ A check in the amount of \$ 120.00 to cover the filing fee is enclosed.					
☐ Payment by credit card. Form PTO-2038 is attached.					
☒ The Director is hereby authorized to charge and credit Deposit Account No. 02-2448 as described below. A duplicate copy of this sheet is enclosed.					
☒ Credit any overpayment.					
☒ Charge any additional filing or application processing fees required under 37 CFR 1.16 and 1.17.					
 Marc S. Weiner Attorney Reg. No.: 32,181 BIRCH, STEWART, KOLASCH & BIRCH, LLP 8110 Gatehouse Road Suite 100 East P.O. Box 747 Falls Church, Virginia 22040-0747 (703) 205-8000				Dated: November 1, 2005	

PATENT APPLICATION FEE DETERMINATION RECORD
Effective October 1, 2000

Application or Docket Number

09/830114

CLAIMS AS FILED - PART I

	(Column 1)	(Column 2)
TOTAL CLAIMS		
FOR	NUMBER FILED	NUMBER EXTRA
TOTAL CHARGEABLE CLAIMS	63 minus 20 =	43
INDEPENDENT CLAIMS	3 minus 3 =	
MULTIPLE DEPENDENT CLAIM PRESENT ☐		

* If the difference in column 1 is less than zero, enter "0" in column 2

CLAIMS AS AMENDED - PART II

11/05

	(Column 1)	(Column 2)	(Column 3)
CLAIMS REMAINING AFTER AMENDMENT		HIGHEST NUMBER PREVIOUSLY PAID FOR	PRESENT EXTRA
Total	63	Minus	63 =
Independent	29	Minus	3 = 26
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM ☐			

	(Column 1)	(Column 2)	(Column 3)
CLAIMS REMAINING AFTER AMENDMENT		HIGHEST NUMBER PREVIOUSLY PAID FOR	PRESENT EXTRA
Total		Minus	=
Independent		Minus	=
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM ☐			

	(Column 1)	(Column 2)	(Column 3)
CLAIMS REMAINING AFTER AMENDMENT		HIGHEST NUMBER PREVIOUSLY PAID FOR	PRESENT EXTRA
Total		Minus	=
Independent		Minus	=
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM ☐			

- * 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.

SMALL ENTITY TYPE ☐

OTHER THAN SMALL ENTITY

RATE	FEE
BASIC FEE	355.00
X\$ 9=	387
X40=	
+135=	
TOTAL	817

RATE	FEE
BASIC FEE	710.00
X\$18=	
X80=	
+270=	
TOTAL	

SMALL ENTITY TYPE ☐

OTHER THAN SMALL ENTITY

RATE	ADDITIONAL FEE
X\$ 9=	
X40=	
+135=	
TOTAL ADDIT. FEE	

RATE	ADDITIONAL FEE
X\$18=	
X80=	
+270=	
TOTAL ADDIT. FEE	

RATE	ADDITIONAL FEE
X\$ 9=	
X40=	
+135=	
TOTAL ADDIT. FEE	

RATE	ADDITIONAL FEE
X\$18=	
X80=	
+270=	
TOTAL ADDIT. FEE	

RATE	ADDITIONAL FEE
X\$ 9=	
X40=	
+135=	
TOTAL ADDIT. FEE	

RATE	ADDITIONAL FEE
X\$18=	
X80=	
+270=	
TOTAL ADDIT. FEE	



UNITED STATES PATENT AND TRADEMARK OFFICE

UNITED STATES DEPARTMENT OF COMMERCE
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P.O. Box 1450
Alexandria, Virginia 22313-1450
www.uspto.gov

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/830,114	06/20/2001	Bruno Besette	4082-0129P	3008
2292	7590	01/17/2006	EXAMINER	
BIRCH STEWART KOLASCH & BIRCH PO BOX 747 FALLS CHURCH, VA 22040-0747			ARMSTRONG, ANGELA A	
			ART UNIT	PAPER NUMBER
			2654	

DATE MAILED: 01/17/2006

Please find below and/or attached an Office communication concerning this application or proceeding.

**Notice of Non-Compliant
Amendment (37 CFR 1.121)**

Application No.

09/830,114

Applicant(s)

BESSETTE ET AL.

Examiner

Angela A. Armstrong

Art Unit

2654

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

The amendment document filed on 01 November 2005 is considered non-compliant because it has failed to meet the requirements of 37 CFR 1.121. In order for the amendment document to be compliant, correction of the following item(s) is required.

THE FOLLOWING MARKED (X) ITEM(S) CAUSE THE AMENDMENT DOCUMENT TO BE NON-COMPLIANT:

- ☐ 1. Amendments to the specification:
 - ☐ A. Amended paragraph(s) do not include markings.
 - ☐ B. New paragraph(s) should not be underlined.
 - ☐ C. Other _____.
- ☐ 2. Abstract:
 - ☐ A. Not presented on a separate sheet. 37 CFR 1.72.
 - ☐ B. Other _____.
- ☐ 3. Amendments to the drawings:
 - ☐ A. The drawings are not properly identified in the top margin as "Replacement Sheet," "New Sheet," or "Annotated Sheet" as required by 37 CFR 1.121(d).
 - ☐ B. The practice of submitting proposed drawing correction has been eliminated. Replacement drawings showing amended figures, without markings, in compliance with 37 CFR 1.84 are required.
 - ☐ C. Other _____.
- ☒ 4. Amendments to the claims:
 - ☐ A. A complete listing of all of the claims is not present.
 - ☐ B. The listing of claims does not include the text of all pending claims (including withdrawn claims)
 - ☒ C. Each claim has not been provided with the proper status identifier, and as such, the individual status of each claim cannot be identified. Note: the status of every claim must be indicated after its claim number by using one of the following status identifiers: (Original), (Currently amended), (Canceled), (Previously presented), (New), (Not entered), (Withdrawn) and (Withdrawn-currently amended).
 - ☐ D. The claims of this amendment paper have not been presented in ascending numerical order.
 - ☒ E. Other: See Continuation Sheet.

For further explanation of the amendment format required by 37 CFR 1.121, see MPEP § 714 and the USPTO website at <http://www.uspto.gov/web/offices/pac/dapp/opla/preognotice/officeflyer.pdf>.

TIME PERIODS FOR FILING A REPLY TO THIS NOTICE:

1. Applicant is given **no new time period** if the non-compliant amendment is an after-final amendment or an amendment filed after allowance. If applicant wishes to resubmit the non-compliant after-final amendment with corrections, the **entire corrected amendment** must be resubmitted within the time period set forth in the final Office action.
2. Applicant is given **one month**, or thirty (30) days, whichever is longer, from the mail date of this notice to supply the **corrected section** of the non-compliant amendment in compliance with 37 CFR 1.121, if the non-compliant amendment is one of the following: a preliminary amendment, a non-final amendment (including a submission for a request for continued examination (RCE) under 37 CFR 1.114), a supplemental amendment filed within a suspension period under 37 CFR 1.103(a) or (c), and an amendment filed in response to a *Quayle* action.

Extensions of time are available under 37 CFR 1.136(a) only if the non-compliant amendment is a non-final amendment or an amendment filed in response to a *Quayle* action.

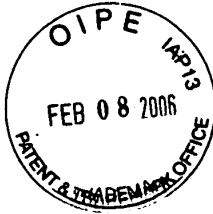
Failure to timely respond to this notice will result in:

Abandonment of the application if the non-compliant amendment is a non-final amendment or an amendment filed in response to a *Quayle* action; or

Non-entry of the amendment if the non-compliant amendment is a preliminary amendment or supplemental amendment.

Angela Armstrong
ANGELA ARMSTRONG
PRIMARY EXAMINER

Continuation of 4(e) Other: The status of claim 16 is indicated as original, however the text of the claim shows markings representative of an amendment to the claim..



TFW

PATENT
Attorney Docket: 4510-0105P

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicants: Bruno BESSETTE et al. Conf.: 3008
Appl No: 09/830,114 Art Unit: 2654
Filed: June 20, 2001 Examiner: A. A. Armstrong
For: A METHOD AND DEVICE FOR ADAPTIVE BANDWIDTH PITCH
 SEARCH IN CODING WIDEBAND SIGNALS

REPLY TO THE NOTICE OF NON-COMPLIANT AMENDMENT

Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Sir:

Responsive to the January 17, 2006 Notice of Non-Compliant Amendment, attached hereto is a corrected version of the "claims" section of the Non-Final Amendment submitted on November 1, 2005.

The present submission is being timely filed within thirty (30) days of the mailing date of the Notice. Accordingly, no extension of time fees should be due.



U.S. Application No. 09/830,114
Docket No. 4510-0105P
Reply filed November 1, 2005
Art Unit: 2654
Page 2 of 29

AMENDMENTS TO THE CLAIMS

Please amend the claims as follows:

1. (CURRENTLY AMENDED) A pitch analysis device for producing an optimal ~~a~~ set of pitch codebook parameters, comprising:

a) at least two signal paths associated to respective sets of pitch codebook parameters, wherein:

i) each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search device; and

ii) at least one of said two signal paths comprises a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of said one signal path; and

b) a selector for comparing the pitch prediction errors calculated in said at least two signal paths, for choosing the signal path having the lowest calculated pitch prediction error and for selecting the set of pitch codebook parameters associated to the chosen signal path.

2. (CURRENTLY AMENDED) A pitch analysis device as defined in claim 1, wherein one of said at least two signal paths comprises no filter for

filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

3. (CURRENTLY AMENDED) A pitch analysis device as defined in claim 1, wherein said signal paths comprise a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same signal path.

4. (CURRENTLY AMENDED) A pitch analysis device as defined in claim 3, wherein the filters of said plurality of signal paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

5. (ORIGINAL) A pitch analysis device as defined in claim 1, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with a weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

6. (ORIGINAL) A pitch analysis device as defined in claim 5, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x' y^{(j)} / \|y^{(j)}\|^2$$

where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

7. (ORIGINAL) A pitch analysis device as defined in claim 1, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said

pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

8. (ORIGINAL) A pitch analysis device as defined in claim 5, wherein:

- a) each of said filters of the plurality of signal paths is identified by a filter index;
- b) said pitch codevector is identified by a pitch codebook index; and
- c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

9. (ORIGINAL) A pitch analysis device as defined in claim 1, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

10. (CURRENTLY AMENDED) A pitch analysis method for producing ~~an optimal~~ a set of pitch codebook parameters, comprising:

a) in at least two signal paths associated to respective sets of pitch codebook parameters, calculating, for each signal path, a pitch prediction error of a pitch codevector from a pitch codebook search device;

b) in at least one of said two signal paths, filtering the pitch codevector before supplying said pitch codevector for calculation of said pitch prediction error of said one signal path; and

c) comparing the pitch prediction errors calculated in said at least two signal paths, choosing the signal path having the lowest calculated pitch prediction error, and selecting the set of pitch codebook parameters associated to the chosen signal path.

11. (CURRENTLY AMENDED) A pitch analysis method as defined in claim 10, wherein, in one of said at least two signal paths, no filtering of the pitch codevector is performed before supplying said pitch codevector to a pitch prediction error calculating device.

12. (CURRENTLY AMENDED) A pitch analysis method as defined in claim 10, wherein said signal paths ~~comprises~~ comprise a plurality of signal

paths and wherein filtering the pitch codevector is performed in each of said plurality of signal paths before supplying said pitch codevector to the pitch prediction error calculating device of the same signal path.

13. (CURRENTLY AMENDED) A pitch analysis method as defined in claim 12, further comprising selecting the filters of said plurality of signal paths from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

14. (ORIGINAL) A pitch analysis method as defined in claim 10, wherein calculating a pitch prediction error in each signal path comprises:

- a) convolving the pitch codevector with a weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;
- b) calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;
- c) multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and
- d) combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

15. (ORIGINAL) A pitch analysis method as defined in claim 14, wherein said pitch gain calculation comprises calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$$

where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

16. (CURRENTLY AMENDED) A pitch analysis method as defined in claim 10, wherein calculating said pitch prediction error, in each signal path, comprises calculating an energy of the corresponding pitch prediction error, and wherein comparing the pitch prediction ~~error~~ errors comprises comparing the energies of said pitch prediction errors of the different signal paths and choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

17. (ORIGINAL) A pitch analysis method as defined in claim 14, wherein:

a) identifying each of said filters of the plurality of signal paths by a filter index;

- b) identifying said pitch codevector by a pitch codebook index; and
- c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

18. (ORIGINAL) A pitch analysis method as defined in claim 10, wherein filtering of the pitch codevector is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

19. (CURRENTLY AMENDED) An encoder having a pitch analysis device as in claim 1 for encoding a wideband input signal, said encoder comprising:

- a) a linear prediction synthesis filter calculator responsive to the wideband signal for producing linear prediction synthesis filter coefficients;
- b) a perceptual weighting filter, responsive to the wideband signal and the linear prediction synthesis filter coefficients, for producing a perceptually weighted signal;
- c) an impulse response generator responsive to said linear prediction synthesis filter coefficients for producing a weighted synthesis filter impulse response signal;

d) a pitch search unit for producing pitch codebook parameters, said pitch search unit comprising:

i) said pitch codebook search device responsive to the perceptually weighted signal and the linear prediction synthesis filter coefficients for producing the pitch codevector and an innovative search target vector; and

ii) said pitch analysis device responsive to the pitch codevector for selecting, from said sets of pitch codebook parameters, the set of pitch codebook parameters associated to the signal path having the lowest calculated pitch prediction error;

e) an innovative codebook search device, responsive to a weighted synthesis filter impulse response signal, and the innovative search target vector, for producing innovative codebook parameters; and

f) a signal forming device for producing an encoded wideband signal comprising the set of pitch codebook parameters associated to the signal path having the lowest pitch prediction error, said innovative codebook parameters, and said linear prediction synthesis filter coefficients.

20. (CURRENTLY AMENDED) An encoder as defined in claim 19, wherein one of said at least two signal paths comprises no filter for filtering the

pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

21. (CURRENTLY AMENDED) An encoder as defined in claim 19, wherein said signal paths ~~comprises~~ comprise a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same signal path.

22. (CURRENTLY AMENDED) An encoder as defined in claim 21, wherein the filters of said plurality of signal paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

23. (ORIGINAL) An encoder as defined in claim 19, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

24. (ORIGINAL) An encoder as defined in claim 23, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$$

where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

25. (ORIGINAL) An encoder as defined in claim 19, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch

prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

26. (ORIGINAL) An encoder as defined in claim 23, wherein:

- a) each of said filters of the plurality of signal paths is identified by a filter index;
- b) said pitch codevector is identified by a pitch codebook index; and
- c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

27. (ORIGINAL) An encoder as defined in claim 19, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

28. (CURRENTLY AMENDED) A cellular communication system for servicing a ~~large~~ geographical area divided into a plurality of cells, comprising:

- a) mobile transmitter/receiver units;
- b) cellular base stations respectively situated in said cells;

c) a control terminal for controlling communication between the cellular base stations; and

d) a bidirectional wireless communication sub-system between each mobile unit situated in one cell and the cellular base station of said one cell, said bidirectional wireless communication sub-system comprising, in both the mobile unit and the cellular base station:

i) a transmitter including an encoder for encoding a wideband signal as recited in claim 19, and a transmission circuit for transmitting the encoded wideband signal; and

ii) a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal.

29. (CURRENTLY AMENDED) A cellular communication system as defined in claim 28, wherein one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

30. (CURRENTLY AMENDED) A cellular communication system as defined in claim 28, wherein said signal paths ~~comprises~~ comprise a plurality

of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same signal path.

31. (CURRENTLY AMENDED) A cellular communication system as defined in claim 30, wherein the filters of said plurality of signal paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

32. (ORIGINAL) A cellular communication system as defined in claim 28, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and the pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

33. (ORIGINAL) A cellular communication system as defined in claim 32, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x' y^{(j)} / \|y^{(j)}\|^2$$

where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

34. (ORIGINAL) A cellular communication system as defined in claim 28, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

35. (ORIGINAL) A cellular communication system as defined in claim 32, wherein:

- a) each of said filters of the plurality of signal paths is identified by a filter index;
- b) said pitch codevector is identified by a pitch codebook index; and
- c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

36. (ORIGINAL) A cellular communication system as defined in claim 28, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

37. (ORIGINAL) A cellular mobile transmitter/receiver unit, comprising:

- a) a transmitter including an encoder for encoding a wideband signal as recited in claim 19 and a transmission circuit for transmitting the encoded wideband signal; and

b) a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal.

38. (CURRENTLY AMENDED) A cellular mobile transmitter/receiver unit as defined in claim 37, wherein one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

39. (CURRENTLY AMENDED) A cellular mobile transmitter/receiver unit as defined in claim 37, wherein said signal paths ~~comprises~~ comprise a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same signal path.

40. (CURRENTLY AMENDED) A cellular mobile transmitter/receiver unit as defined in claim 39, wherein the filters of said plurality of signal paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

41. (ORIGINAL) A cellular mobile transmitter/receiver unit as defined in claim 37, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

42. (ORIGINAL) A cellular mobile transmitter/receiver unit as defined in claim 41, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x' y^{(j)} / \|y^{(j)}\|^2$$

where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

43. (ORIGINAL) A cellular mobile transmitter/receiver unit as defined in claim 37, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

44. (ORIGINAL) A cellular mobile transmitter/receiver unit as defined in claim 41, wherein:

- a) each of said filters of the plurality of signal paths is identified by a filter index;
- b) said pitch codevector is identified by a pitch codebook index; and
- c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

45. (ORIGINAL) A cellular mobile transmitter/receiver unit as defined in claim 37, wherein said filter is integrated in an interpolation filter of said

pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

46. (CURRENTLY AMENDED) A ~~cellular~~ network element, comprising:

a) a transmitter including an encoder for encoding a wideband signal as recited in claim 19 and a transmission circuit for transmitting the encoded wideband signal ; and

~~b) a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal.~~

47. (CURRENTLY AMENDED) A ~~cellular~~ network element as defined in claim 46, wherein one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

48. (CURRENTLY AMENDED) A ~~cellular~~ network element as defined in claim 46, wherein said signal paths ~~comprises~~ comprise a plurality of signal paths each provided with a filter for filtering the pitch codevector before

supplying said pitch codevector to the pitch prediction error calculating device of the same ~~signal~~ path.

49. (CURRENTLY AMENDED) A ~~cellular~~ network element as defined in claim 48, wherein the filters of said plurality of signal paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

50. (CURRENTLY AMENDED) A ~~cellular~~ network element as defined in claim 46, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

51. (CURRENTLY AMENDED) A ~~cellular~~ network element as defined in claim 50, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x' y^{(j)} / \|y^{(j)}\|^2$$

where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

52. (CURRENTLY AMENDED) A ~~cellular~~ network element as defined in claim 46, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

53. (CURRENTLY AMENDED) A ~~cellular~~ network element as defined in claim 50, wherein:

- a) each of said filters of the plurality of signal paths is identified by a filter index;
- b) said pitch codevector is identified by a pitch codebook index; and
- c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

54. (CURRENTLY AMENDED) A ~~cellular~~ network element as defined in claim 46, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

55. (CURRENTLY AMENDED) In a cellular communication system for servicing a ~~large~~ geographical area divided into a plurality of cells, comprising: mobile transmitter/receiver units, cellular base stations ~~—~~ respectively situated in said cells; and a control terminal for controlling communication between the cellular base stations;

a bidirectional wireless communication sub-system between each mobile unit situated in one cell and the cellular base station of said one cell, said

bidirectional wireless communication sub-system comprising, in both the mobile unit and the cellular base station:

a) a transmitter including an encoder for encoding a wideband signal as recited in claim 19, and a transmission circuit for transmitting the encoded wideband signal; and

b) a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal.

56. (CURRENTLY AMENDED) A bidirectional wireless communication sub-system as defined in claim 55, wherein one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

57. (CURRENTLY AMENDED) A bidirectional wireless communication sub-system as defined in claim 55, wherein said signal paths ~~comprises~~ comprise a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same signal path.

58. (CURRENTLY AMENDED) A bidirectional wireless communication sub-system as defined in claim 57, wherein the filters of said plurality of signal paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

59. (ORIGINAL) A bidirectional wireless communication sub-system as defined in claim 55, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

60. (ORIGINAL) A bidirectional wireless communication sub-system as defined in claim 59, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$$

where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

61. (ORIGINAL) A bidirectional wireless communication sub-system as defined in claim 55, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

62. (ORIGINAL) A bidirectional wireless communication sub-system as defined in claim 59, wherein:

- a) each of said filters of the plurality of signal paths is identified by a filter index;
- b) said pitch codevector is identified by a pitch codebook index; and
- c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

63. (ORIGINAL) A bidirectional wireless communication sub-system as defined in claim 55, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

U.S. Application No. 09/830,114
Docket No. 4510-0105P
Reply filed November 1, 2005
Art Unit: 2654
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REMARKS

Because the Rule 111 Amendment was filed as part of a non-final amendment, only the corrected section is supplied in this Reply. Applicants respectfully request that all of the claims, including the correctly resubmitted claim 16 be accepted for examination.

All objections and rejections raised in the Notice having been addressed, it is respectfully submitted that the present application is in condition for allowance. Should there be any outstanding matters that need to be resolved, the Examiner is respectfully requested to contact Hyung Sohn (Reg. No. 44,346), to conduct an interview in an effort to expedite prosecution in connection with the present application.

Dated:

FEB 8 2006

Respectfully submitted,

BIRCH, STEWART, KOLASCH &, BIRCH, LLP

By:



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EAST Search History

Ref #	Hits	Search Query	DBs	Default Operator	Plurals	Time Stamp
S16	2006	704/219,220,223.ccls.	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/05/31 07:37
S17	0	("2005/0096901").URPN.	USPAT	OR	ON	2006/05/31 07:40
S18	0	("2004/0215450").URPN.	USPAT	OR	ON	2006/05/31 07:41
S19	16	("4220819" "4797925" "4817157" "5271089" "5274741" "5353373" "5371853" "5451951" "5621852" "5657418" "5787390" "5845244" "6148282" "6161086" "6240382" "6389388"). PN. OR ("6763330").URPN.	US-PGPUB; USPAT; USOCR	OR	ON	2006/05/31 08:02
S5	54	S3 and S4	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/05/31 08:03
S20	210	pitch with prediction with error	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/05/31 08:03
S21	373	pitch same codebook same search	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/05/31 08:03
S22	59	S20 and S21	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/05/31 08:03
S23	0	("2002/0103638").URPN.	USPAT	OR	ON	2006/05/31 08:03
S6	14	("4932061" "5307441" "5359696" "5444816" "5519807" "5664055" "5699482" "5701392" "5754976" "5963898" "6006174" "6064962" "6192334" "6449590").PN. OR ("6807524"). URPN.	US-PGPUB; USPAT; USOCR	OR	ON	2006/05/31 08:06
S7	150	filter\$5 same codevector	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/05/31 08:06
S24	15	("4932061" "5307441" "5359696" "5444816" "5519807" "5664055" "5699482" "5701392" "5754976" "5963898" "6006174" "6064962" "6192334" "6449590").PN. OR ("6807524"). URPN.	US-PGPUB; USPAT; USOCR	OR	ON	2006/05/31 08:06
S25	160	filter\$5 same codevector	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/05/31 08:06
S26	0	("2005/0108005").URPN.	USPAT	OR	ON	2006/05/31 08:13

EAST S arch History

S27	15	("4932061" "5307441" "5359696" "5444816" "5519807" "5664055" "5699482" "5701392" "5754976" "5963898" "6006174" "6064962" "6192334" "6449590").PN. OR ("6807524").URPN.	US-PGPUB; USPAT; USOCR	OR	ON	2006/05/31 08:15
S11	291	pitch with predict\$ with error	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/05/31 08:17
S28	320	pitch with predict\$ with error	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/05/31 08:17
S29	69	S21 and S28	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/05/31 08:18
S14	8	S12 same S13	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/05/31 08:19
S30	50	convol\$5 with pitch with \$4vector	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/05/31 08:19
S13	968	weight\$ with synthesi\$ with filter	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/05/31 08:20
S31	1058	weight\$3 with synthesi\$ with filter	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/05/31 08:20
S32	11	S30 same S31	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/05/31 08:20
S33	4623	pitch with analy\$4	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/06/01 09:36
S34	1050	(pitch adaptive) with codebook	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/06/01 09:36
S35	209	S33 same S34	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/06/01 09:36

EAST Search History

S36	415	pitch with prediction with (error residual)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/06/01 09:36
S37	89	S34 same S36	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/06/01 09:37
S38	15	("4932061" "5307441" "5359696" "5444816" "5519807" "5664055" "5699482" "5701392" "5754976" "5963898" "6006174" "6064962" "6192334" "6449590").PN. OR ("6807524"). URPN.	US-PGPUB; USPAT; USOCR	OR	ON	2006/06/01 09:46
S39	7	("5235669" "5444816" "5450449" "5699482" "5701392" "5754976" "5819213").PN. OR ("6795805").URPN.	US-PGPUB; USPAT; USOCR	OR	ON	2006/06/01 09:49
S40	1630	short with term with predict\$4	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/06/02 11:34
S41	2720	long with term with predict\$4	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/06/02 11:34
S42	754	S40 same S41	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/06/02 11:34
S43	275	S42 and "704"/\$.ccls.	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/06/02 11:34
S44	619	pitch with predict\$4 with (error residual)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/06/02 11:35
S45	53	S42 same S44	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2006/06/02 11:35



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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/830,114	06/20/2001	Bruno Bessette	4082-0129P	3008
2292	7590	06/08/2006	EXAMINER	
BIRCH STEWART KOLASCH & BIRCH PO BOX 747 FALLS CHURCH, VA 22040-0747			ARMSTRONG, ANGELA A	
			ART UNIT	PAPER NUMBER
			2626	

DATE MAILED: 06/08/2006

Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Applicant No.	Applicant(s)	
	09/830,114	BESSETTE ET AL.	
	Examiner	Art Unit	
	Angela A. Armstrong	2626	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

- 1) ☒ Responsive to communication(s) filed on 08 February 2006.
- 2a) ☐ This action is **FINAL**. 2b) ☒ This action is non-final.
- 3) ☐ 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.

Disposition of Claims

- 4) ☒ Claim(s) 1-63 is/are pending in the application.
- 4a) Of the above claim(s) _____ is/are withdrawn from consideration.
- 5) ☐ Claim(s) _____ is/are allowed.
- 6) ☒ Claim(s) 1-63 is/are rejected.
- 7) ☐ Claim(s) _____ is/are objected to.
- 8) ☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

- 9) ☐ The specification is objected to by the Examiner.
- 10) ☐ The drawing(s) filed on _____ is/are: a) ☐ accepted or b) ☐ objected to by the Examiner.
Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).
Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).
- 11) ☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

- 12) ☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).
- a) ☐ All b) ☐ Some * c) ☐ None of:
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)).
- * See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☐ Notice of References Cited (PTO-892)	4) ☐ Interview Summary (PTO-413)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)	Paper No(s)/Mail Date: _____.
3) ☐ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08)	5) ☐ Notice of Informal Patent Application (PTO-152)
Paper No(s)/Mail Date: _____.	6) ☐ Other: _____.

DETAILED ACTION

Claim Rejections - 35 USC § 101

35 U.S.C. 101 reads as follows:

Whoever invents or discovers any new and useful process, machine, manufacture, or composition of matter, or any new and useful improvement thereof, may obtain a patent therefor, subject to the conditions and requirements of this title.

Claims 1-63 are rejected under 35 U.S.C. 101 because the claimed invention is directed to non-statutory subject matter.

Claims 1-63 define non-statutory processed because they merely manipulate an abstract idea. The claimed process, a series of steps to be performed by a computer, amounts to a manipulation of an abstract idea since the process fails to provide any pre- or post- computer process activity.

The claims fail to include limitations of functional descriptive material that can impart functionality when employed as computer components so as to yield a useful, tangible, concrete result.

Applicant should note, however, that claims directed to speech or audio signal processing, would be considered to be statutory subject matter. For example, the requirement of the measurements of physical objects or activities to be transformed outside of the computer into computer data (In re Gelnovatch, 595 F.2d 32, 41 n.7, 201 USPQ 136, 145 n.7 (CCPA 1979) (data- gathering step did not measure physical phenomenon); Arrhythmia, 958 F.2d at 1056, 22 USPQ2d at 1036), where the data comprises signals corresponding to physical objects or activities external to the computer system, and where the process causes a physical transformation of the signals which are intangible representations of the physical objects

or activities. Schrader, 22 F.3d at 294, 30 USPQ2d at 1459 citing with approval Arrhythmia, 958 F.2d at 1058-59, 22 USPQ2d at 1037-38; Abele, 684 F.2d at 909, 214 USPQ at 688; In re Taner, 681 F.2d 787, 790, 214 USPQ 678, 681 (CCPA 1982).

Examples of this type of claimed statutory process include the following:

- A method of using a computer processor to analyze electrical signals and data representative of human cardiac activity by converting the signals to time segments, applying the time segments in reverse order to a high pass filter means, using the computer processor to determine the amplitude of the high pass filter's output, and using the computer processor to compare the value to a predetermined value. In this example the data is an intangible representation of physical activity, i.e., human cardiac activity. The transformation occurs when heart activity is measured and an electrical signal is produced. This process has real world value in predicting vulnerability to ventricular tachycardia immediately after a heart attack.

- A method of using a computer processor to receive data representing Computerized Axial Tomography ("CAT") scan images of a patient, performing a calculation to determine the difference between a local value at a data point and an average value of the data in a region surrounding the point, and displaying the difference as a gray scale for each point in the image, and displaying the resulting image. In this example the data is an intangible representation of a physical object, i.e., portions of the anatomy of a patient. The transformation occurs when the condition of the human body is measured with X-rays and the X-rays are converted into electrical digital signals that represent the condition of the human body. The real world value of the invention lies in creating a new CAT scan image of body tissue without the presence of bones.

- A method of using a computer processor to conduct seismic exploration, by imparting spherical seismic energy waves into the earth from a seismic source, generating a plurality of reflected signals in response to the seismic energy waves at a set of receiver positions in an array, and summing the reflection signals to produce a signal simulating the reflection response of the earth to the seismic energy. In this example, the electrical signals processed by the computer represent reflected seismic energy. The transformation occurs by converting the spherical seismic energy waves into electrical signals, which provide a geophysical representation of formations below the earth's surface. Geophysical exploration of formations below the surface of the earth has real world value.

Examples of claimed processes that independently limit the claimed invention to safe harbor include:

- a method of conducting seismic exploration which requires generating and manipulating signals from seismic energy waves before "summing" the values represented by the signals (Taner, 681 F.2d at 788, 214 USPQ at 679); and

- a method of displaying X-ray attenuation data as a signed gray scale signal in a "field" using a particular algorithm, where the antecedent steps require generating the data using a particular machine (e.g., a computer tomography scanner). Abele, 684 F.2d at 908, 214 USPQ at 687 ("The specification indicates that such attenuation data is available only when an X-ray beam is produced by a CAT scanner, passed through an object, and detected upon its exit. Only after these steps have been completed is the algorithm performed, and the resultant modified data displayed in the required format.").

Examples of claimed processes that do not limit the claimed invention to pre-computing safe harbor include:

- "perturbing" the values of a set of process inputs, where the subject matter "perturbed" was a number and the act of "perturbing" consists of substituting the numerical values of variables (Gelnovatch, 595 F.2d at 41 n.7, 201 USPQ at 145 n.7 ("Appellants' claimed step of perturbing the values of a set of process inputs (step 3), in addition to being a mathematical operation, appears to be a data-gathering step of the type we have held insufficient to change a nonstatutory method of calculation into a statutory process.... In this instance, the perturbed process inputs are not even measured values of physical phenomena, but are instead derived by numerically changing the values in the previous set of process inputs.")); and, selecting a set of arbitrary measurement point values (Sarkar, 588 F.2d at 1331, 200 USPQ at 135). If a claim does not clearly fall into one or both of the safe harbors, the claim may still be statutory if it is limited to a practical application in the technological arts.

Response to Arguments

Applicant's arguments with respect to claims 1-63 have been considered but are moot in view of the new ground(s) of rejection.

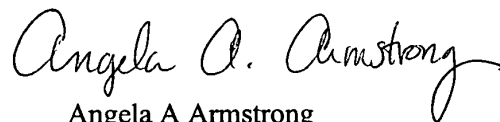
Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Angela A. Armstrong whose telephone number is 571-272-7598. The examiner can normally be reached on Monday-Thursday 11:30-8:00 PM.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, David Hudspeth can be reached on 571-272-7843. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Please note the change in art unit designation for the examiner from old art unit "2654" to new art unit "2626."

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see <http://pair-direct.uspto.gov>. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.



Angela A Armstrong
Primary Examiner
Art Unit 2626

AAA
June 5, 2006

Index of Claims 	Application/Counter IN .		Applicant(s)/Patent number	
	09/830,114		BESSETTE ET AL.	
	Examiner		Art Unit	
	Angela A. Armstrong		2626	

✓ Rejected	— (Through numeral) Cancelled	N Non-Entered	A Appraised
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Claim		Date									
Final	Original	6/5/06									
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Search Notes 	Application/Contr. No. 09/830,114 Examiner Angela A. Armstrong	Applicant(s)/Patent under R. examination BESSETTE ET AL. Art Unit 2626

SEARCHED			
Class	Subclass	Date	Examiner

INTERFERENCE SEARCHED			
Class	Subclass	Date	Examiner

SEARCH NOTES (INCLUDING SEARCH STRATEGY)		
	DATE	EXMR
EAST text search	5/31/2006	AAA
EAST text search	6/1/2006	AAA
EAST text search	6/2/2006	AAA



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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/830,114	06/20/2001	Bruno Bessette	4082-0129P	3008
2292	7590	09/06/2006	EXAMINER	
BIRCH STEWART KOLASCH & BIRCH PO BOX 747 FALLS CHURCH, VA 22040-0747			ARMSTRONG, ANGELA A	
			ART UNIT	PAPER NUMBER
			2626	

DATE MAILED: 09/06/2006

Please find below and/or attached an Office communication concerning this application or proceeding.

Summary of Record of Interview Requirements

Manual of Patent Examining Procedure (MPEP), Section 713.04, Substance of Interview Must be Made of Record

A complete written statement as to the substance of any face-to-face, video conference, or telephone interview with regard to an application must be made of record in the application whether or not an agreement with the examiner was reached at the interview.

Title 37 Code of Federal Regulations (CFR) § 1.133 Interviews

Paragraph (b)

In every instance where reconsideration is requested in view of an interview with an examiner, a complete written statement of the reasons presented at the interview as warranting favorable action must be filed by the applicant. An interview does not remove the necessity for reply to Office action as specified in §§ 1.111, 1.135. (35 U.S.C. 132)

37 CFR §1.2 Business to be transacted in writing.

All business with the Patent or Trademark Office should be transacted in writing. The personal attendance of applicants or their attorneys or agents at the Patent and Trademark Office is unnecessary. The action of the Patent and Trademark Office will be based exclusively on the written record in the Office. No attention will be paid to any alleged oral promise, stipulation, or understanding in relation to which there is disagreement or doubt.

The action of the Patent and Trademark Office cannot be based exclusively on the written record in the Office if that record is itself incomplete through the failure to record the substance of interviews.

It is the responsibility of the applicant or the attorney or agent to make the substance of an interview of record in the application file, unless the examiner indicates he or she will do so. It is the examiner's responsibility to see that such a record is made and to correct material inaccuracies which bear directly on the question of patentability.

Examiners must complete an Interview Summary Form for each interview held where a matter of substance has been discussed during the interview by checking the appropriate boxes and filling in the blanks. Discussions regarding only procedural matters, directed solely to restriction requirements for which interview recodation is otherwise provided for in Section 812.01 of the Manual of Patent Examining Procedure, or pointing out typographical errors or unreadable script in Office actions or the like, are excluded from the interview recodation procedures below. Where the substance of an interview is completely recorded in an Examiners Amendment, no separate Interview Summary Record is required.

The Interview Summary Form shall be given an appropriate Paper No., placed in the right hand portion of the file, and listed on the "Contents" section of the file wrapper. In a personal interview, a duplicate of the Form is given to the applicant (or attorney or agent) at the conclusion of the interview. In the case of a telephone or video-conference interview, the copy is mailed to the applicant's correspondence address either with or prior to the next official communication. If additional correspondence from the examiner is not likely before an allowance or if other circumstances dictate, the Form should be mailed promptly after the interview rather than with the next official communication.

The Form provides for recodation of the following information:

- Application Number (Series Code and Serial Number)
- Name of applicant
- Name of examiner
- Date of interview
- Type of interview (telephonic, video-conference, or personal)
- Name of participant(s) (applicant, attorney or agent, examiner, other PTO personnel, etc.)
- An indication whether or not an exhibit was shown or a demonstration conducted
- An identification of the specific prior art discussed
- An indication whether an agreement was reached and if so, a description of the general nature of the agreement (may be by attachment of a copy of amendments or claims agreed as being allowable). Note: Agreement as to allowability is tentative and does not restrict further action by the examiner to the contrary.
- The signature of the examiner who conducted the interview (if Form is not an attachment to a signed Office action)

It is desirable that the examiner orally remind the applicant of his or her obligation to record the substance of the interview of each case. It should be noted, however, that the Interview Summary Form will not normally be considered a complete and proper recodation of the interview unless it includes, or is supplemented by the applicant or the examiner to include, all of the applicable items required below concerning the substance of the interview.

A complete and proper recodation of the substance of any interview should include at least the following applicable items:

- 1) A brief description of the nature of any exhibit shown or any demonstration conducted,
- 2) an identification of the claims discussed,
- 3) an identification of the specific prior art discussed,
- 4) an identification of the principal proposed amendments of a substantive nature discussed, unless these are already described on the Interview Summary Form completed by the Examiner,
- 5) a brief identification of the general thrust of the principal arguments presented to the examiner,
(The identification of arguments need not be lengthy or elaborate. A verbatim or highly detailed description of the arguments is not required. The identification of the arguments is sufficient if the general nature or thrust of the principal arguments made to the examiner can be understood in the context of the application file. Of course, the applicant may desire to emphasize and fully describe those arguments which he or she feels were or might be persuasive to the examiner.)
- 6) a general indication of any other pertinent matters discussed, and
- 7) if appropriate, the general results or outcome of the interview unless already described in the Interview Summary Form completed by the examiner.

Examiners are expected to carefully review the applicant's record of the substance of an interview. If the record is not complete and accurate, the examiner will give the applicant an extendable one month time period to correct the record.

Examiner to Check for Accuracy

If the claims are allowable for other reasons of record, the examiner should send a letter setting forth the examiner's version of the statement attributed to him or her. If the record is complete and accurate, the examiner should place the indication, "Interview Record OK" on the paper recording the substance of the interview along with the date and the examiner's initials.



Attorney Docket: 4510-0105P

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicants: Bruno BESSETTE et al. Conf.: 3008
Appl No: 09/830,114 Art Unit: 2626
Filed: June 20, 2001 Examiner: A. Armstrong
For: METHOD AND DEVICE FOR ADAPTIVE BANDWIDTH PITCH
SEARCH IN CODING WIDEBAND SIGNALS

REPLY UNDER 37 C.F.R. § 1.111

Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Date: November 8, 2006

Sir:

In reply to the Examiner's Non-Final Office Action dated June 8, 2006, the following amendments and remarks are respectfully submitted in connection with the above-identified application as follows:

- Amendments to the claims;
- Remarks/Arguments.



U.S. Application No. 09/830,114
Docket No. 4510-0105P
Art Unit: 2626
Page 2 of 30

AMENDMENTS TO THE CLAIMS

Please amend the claims as follows:

1. (CURRENTLY AMENDED) A pitch analysis device for producing a set of pitch codebook parameters, comprising:

a pitch codebook search device configured to generate a pitch code vector based on a digitized input audio data, wherein said digitized input audio data represents an input audio signal that has been sampled and digitized;

a) at least two signal paths associated to respective sets of pitch codebook parameters representative of said digitized input audio data, wherein:

i) each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of ~~a~~a said pitch codevector from ~~a~~a said pitch codebook search device; and

ii) at least one of said at least two signal paths comprises a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of said at least one signal path; and

b) a selector for comparing the pitch prediction errors calculated in said at least two signal paths, for choosing the signal path having the lowest calculated pitch prediction error and for selecting the set of pitch codebook parameters associated to the chosen signal path.

2. (PREVIOUSLY PRESENTED) A pitch analysis device as defined in claim 1, wherein one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

3. (PREVIOUSLY PRESENTED) A pitch analysis device as defined in claim 1, wherein said signal paths comprise a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same signal path.

4. (PREVIOUSLY PRESENTED) A pitch analysis device as defined in claim 3, wherein the filters of said plurality of signal paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

5. (ORIGINAL) A pitch analysis device as defined in claim 1, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with a weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

6. (ORIGINAL) A pitch analysis device as defined in claim 5, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$$

where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

7. (ORIGINAL) A pitch analysis device as defined in claim 1, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said

pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

8. (ORIGINAL) A pitch analysis device as defined in claim 5, wherein:

a) each of said filters of the plurality of signal paths is identified by a filter index;

b) said pitch codevector is identified by a pitch codebook index; and

c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

9. (ORIGINAL) A pitch analysis device as defined in claim 1, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

10. (CURRENTLY AMENDED) A pitch analysis method for producing a set of pitch codebook parameters, comprising:

generating a pitch code vector from a pitch codebook search device based on a digitized input audio data, wherein said digitized input audio data represents an input audio signal that has been sampled and digitized;

a) in at least two signal paths associated to respective sets of pitch codebook parameters representative of said digitized input audio data, calculating, for each signal path, a pitch prediction error of a said pitch codevector from a said pitch codebook search device;

b) in at least one of said at least two signal paths, filtering the pitch codevector before supplying said pitch codevector for calculation of said pitch prediction error of said at least one signal path; and

c) comparing the pitch prediction errors calculated in said at least two signal paths, choosing the signal path having the lowest calculated pitch prediction error, and selecting the set of pitch codebook parameters associated to the chosen signal path.

11. (PREVIOUSLY PRESENTED) A pitch analysis method as defined in claim 10, wherein, in one of said at least two signal paths, no filtering of the pitch codevector is performed before supplying said pitch codevector to a pitch prediction error calculating device.

12. (PREVIOUSLY PRESENTED) A pitch analysis method as defined in claim 10, wherein said signal paths comprise a plurality of signal paths and wherein filtering the pitch codevector is performed in each of said plurality of

signal paths before supplying said pitch codevector to the pitch prediction error calculating device of the same signal path.

13. (PREVIOUSLY PRESENTED) A pitch analysis method as defined in claim 12, further comprising selecting the filters of said plurality of signal paths from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

14. (ORIGINAL) A pitch analysis method as defined in claim 10, wherein calculating a pitch prediction error in each signal path comprises:

a) convolving the pitch codevector with a weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;

c) multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

15. (ORIGINAL) A pitch analysis method as defined in claim 14, wherein said pitch gain calculation comprises calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x' y^{(j)} / \|y^{(j)}\|^2$$

where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

16. (PREVIOUSLY PRESENTED) A pitch analysis method as defined in claim 10, wherein calculating said pitch prediction error, in each signal path, comprises calculating an energy of the corresponding pitch prediction error, and wherein comparing the pitch prediction errors comprises comparing the energies of said pitch prediction errors of the different signal paths and choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

17. (ORIGINAL) A pitch analysis method as defined in claim 14, wherein:

a) identifying each of said filters of the plurality of signal paths by a filter index;

- b) identifying said pitch codevector by a pitch codebook index; and
- c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

18. (ORIGINAL) A pitch analysis method as defined in claim 10, wherein filtering of the pitch codevector is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

19. (PREVIOUSLY PRESENTED) An encoder having a pitch analysis device as in claim 1 for encoding a wideband input signal, said encoder comprising:

- a) a linear prediction synthesis filter calculator responsive to the wideband signal for producing linear prediction synthesis filter coefficients;
- b) a perceptual weighting filter, responsive to the wideband signal and the linear prediction synthesis filter coefficients, for producing a perceptually weighted signal;
- c) an impulse response generator responsive to said linear prediction synthesis filter coefficients for producing a weighted synthesis filter impulse response signal;

d) a pitch search unit for producing pitch codebook parameters, said pitch search unit comprising:

i) said pitch codebook search device responsive to the perceptually weighted signal and the linear prediction synthesis filter coefficients for producing the pitch codevector and an innovative search target vector; and

ii) said pitch analysis device responsive to the pitch codevector for selecting, from said sets of pitch codebook parameters, the set of pitch codebook parameters associated to the signal path having the lowest calculated pitch prediction error;

e) an innovative codebook search device, responsive to a weighted synthesis filter impulse response signal, and the innovative search target vector, for producing innovative codebook parameters; and

f) a signal forming device for producing an encoded wideband signal comprising the set of pitch codebook parameters associated to the signal path having the lowest pitch prediction error, said innovative codebook parameters, and said linear prediction synthesis filter coefficients.

20. (PREVIOUSLY PRESENTED) An encoder as defined in claim 19, wherein one of said at least two signal paths comprises no filter for filtering the

pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

21. (PREVIOUSLY PRESENTED) An encoder as defined in claim 19, wherein said signal paths comprise a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same signal path.

22. (PREVIOUSLY PRESENTED) An encoder as defined in claim 21, wherein the filters of said plurality of signal paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

23. (ORIGINAL) An encoder as defined in claim 19, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

24. (ORIGINAL) An encoder as defined in claim 23, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x' y^{(j)} / \|y^{(j)}\|^2$$

where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

25. (ORIGINAL) An encoder as defined in claim 19, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal

path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

26. (ORIGINAL) An encoder as defined in claim 23, wherein:

- a) each of said filters of the plurality of signal paths is identified by a filter index;
- b) said pitch codevector is identified by a pitch codebook index; and
- c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

27. (ORIGINAL) An encoder as defined in claim 19, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

28. (PREVIOUSLY PRESENTED) A cellular communication system for servicing a geographical area divided into a plurality of cells, comprising:

- a) mobile transmitter/receiver units;
- b) cellular base stations respectively situated in said cells;
- c) a control terminal for controlling communication between the cellular base stations; and

d) a bidirectional wireless communication sub-system between each mobile unit situated in one cell and the cellular base station of said one cell, said bidirectional wireless communication sub-system comprising, in both the mobile unit and the cellular base station:

i) a transmitter including an encoder for encoding a wideband signal as recited in claim 19, and a transmission circuit for transmitting the encoded wideband signal; and

ii) a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal.

29. (PREVIOUSLY PRESENTED) A cellular communication system as defined in claim 28, wherein one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

30. (PREVIOUSLY PRESENTED) A cellular communication system as defined in claim 28, wherein said signal paths comprise a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same signal path.

31. (PREVIOUSLY PRESENTED) A cellular communication system as defined in claim 30, wherein the filters of said plurality of signal paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

32. (ORIGINAL) A cellular communication system as defined in claim 28, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and the pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

33. (ORIGINAL) A cellular communication system as defined in claim 32, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^T y^{(j)} / \|y^{(j)}\|^2$$

where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

34. (ORIGINAL) A cellular communication system as defined in claim 28, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

35. (ORIGINAL) A cellular communication system as defined in claim 32, wherein:

a) each of said filters of the plurality of signal paths is identified by a filter index;

- b) said pitch codevector is identified by a pitch codebook index; and
- c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

36. (ORIGINAL) A cellular communication system as defined in claim 28, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

37. (ORIGINAL) A cellular mobile transmitter/receiver unit, comprising:

- a) a transmitter including an encoder for encoding a wideband signal as recited in claim 19 and a transmission circuit for transmitting the encoded wideband signal; and
- b) a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal.

38. (PREVIOUSLY PRESENTED) A cellular mobile transmitter/receiver unit as defined in claim 37, wherein one of said at least two signal paths

comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

39. (PREVIOUSLY PRESENTED) A cellular mobile transmitter/receiver unit as defined in claim 37, wherein said signal paths comprise a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same signal path.

40. (PREVIOUSLY PRESENTED) A cellular mobile transmitter/receiver unit as defined in claim 39, wherein the filters of said plurality of signal paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

41. (ORIGINAL) A cellular mobile transmitter/receiver unit as defined in claim 37, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

42. (ORIGINAL) A cellular mobile transmitter/receiver unit as defined in claim 41, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x' y^{(j)} / \|y^{(j)}\|^2$$

where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

43. (ORIGINAL) A cellular mobile transmitter/receiver unit as defined in claim 37, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch

prediction error the signal path having the lowest calculated energy of the pitch prediction error.

44. (ORIGINAL) A cellular mobile transmitter/receiver unit as defined in claim 41, wherein:

a) each of said filters of the plurality of signal paths is identified by a filter index;

b) said pitch codevector is identified by a pitch codebook index; and

c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

45. (ORIGINAL) A cellular mobile transmitter/receiver unit as defined in claim 37, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

46. (PREVIOUSLY PRESENTED) A network element, comprising:

a transmitter including an encoder for encoding a wideband signal as recited in claim 19 and a transmission circuit for transmitting the encoded wideband signal.

47. (PREVIOUSLY PRESENTED) A network element as defined in claim 46, wherein one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

48. (PREVIOUSLY PRESENTED) A network element as defined in claim 46, wherein said signal paths comprise a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same path.

49. (PREVIOUSLY PRESENTED) A network element as defined in claim 48, wherein the filters of said plurality of paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

50. (PREVIOUSLY PRESENTED) A network element as defined in claim 46, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

51. (PREVIOUSLY PRESENTED) A network element as defined in claim 50, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x' y^{(j)} / \|y^{(j)}\|^2$$

where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

52. (PREVIOUSLY PRESENTED) A network element as defined in claim 46, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the

energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

53. (PREVIOUSLY PRESENTED) A network element as defined in claim 50, wherein:

- a) each of said filters of the plurality of signal paths is identified by a filter index;
- b) said pitch codevector is identified by a pitch codebook index; and
- c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

54. (PREVIOUSLY PRESENTED) A network element as defined in claim 46, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

55. (PREVIOUSLY PRESENTED) In a cellular communication system for servicing a geographical area divided into a plurality of cells, comprising: mobile transmitter/receiver units, cellular base stations respectively situated in said cells; and a control terminal for controlling communication between the

cellular base stations; a bidirectional wireless communication sub-system between each mobile unit situated in one cell and the cellular base station of said one cell, said bidirectional wireless communication sub-system comprising, in both the mobile unit and the cellular base station:

a) a transmitter including an encoder for encoding a wideband signal as recited in claim 19, and a transmission circuit for transmitting the encoded wideband signal; and

b) a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal.

56. (PREVIOUSLY PRESENTED) A bidirectional wireless communication sub-system as defined in claim 55, wherein one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.

57. (PREVIOUSLY PRESENTED) A bidirectional wireless communication sub-system as defined in claim 55, wherein said signal paths comprise a plurality of signal paths each provided with a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of the same signal path.

58. (PREVIOUSLY PRESENTED) A bidirectional wireless communication sub-system as defined in claim 57, wherein the filters of said plurality of signal paths are selected from the group consisting of low-pass and band-pass filters, and wherein said filters have different frequency responses.

59. (ORIGINAL) A bidirectional wireless communication sub-system as defined in claim 55, wherein each pitch prediction error calculating device comprises:

a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;

c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector; and

d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

60. (ORIGINAL) A bidirectional wireless communication sub-system as defined in claim 59, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x' y^{(j)} / \|y^{(j)}\|^2$$

where $j = 0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

61. (ORIGINAL) A bidirectional wireless communication sub-system as defined in claim 55, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error.

62. (ORIGINAL) A bidirectional wireless communication sub-system as defined in claim 59, wherein:

- a) each of said filters of the plurality of signal paths is identified by a filter index;
- b) said pitch codevector is identified by a pitch codebook index; and
- c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

63. (ORIGINAL) A bidirectional wireless communication sub-system as defined in claim 55, wherein said filter is integrated in an interpolation filter of said pitch codebook search device, said interpolation filter being used to produce a sub-sample version of said pitch codevector.

REMARKS

Favorable reconsideration and allowance of the present application are respectfully requested in view of the following remarks. Claims 1-63 remain pending. Claims 1 and 10 are independent.

DISCUSSIONS CONDUCTED

Applicants thank the Examiner for conducting discussion with Applicants' representative between August 2 and September 1, 2006. The amendments to independent claims 1 and 10 reflect the results of the discussions conducted.

§ 101 REJECTION

Claims 1-63 stand rejected under 35 U.S.C. § 101 for allegedly being directed to non-statutory subject matter. Independent claims 1 and 10 are amended based on the discussions conducted between the Examiner and Applicants' representative. The Interview Summary dated August 31, 2006 indicates that the amendments address the outstanding Section 101 issues. Accordingly, Applicants respectfully request that the rejection of claims 1-63 under § 101 be withdrawn.

CONCLUSION

All objections and rejections raised in the Office Action having been addressed, it is respectfully submitted that the present application is in condition for allowance. Should there be any outstanding matters that need to be resolved, the Examiner is respectfully requested to contact Hyung Sohn (Reg. No. 44,346), to conduct an interview in an effort to expedite prosecution in connection with the present application.

Pursuant to the provisions of 37 C.F.R. § 1.17 and 1.136(a), the Applicants hereby petition for an extension of two (2) month to November 8, 2006 in which to file a reply to the Office Action. The required fee of \$450.00 is enclosed herewith.

U.S. Application No. 09/830,114

Docket No. 4510-0105P

Art Unit: 2626

Page 30 of 30

If necessary, the Commissioner is hereby authorized in this, concurrent, and future replies, to charge payment or credit any overpayment to Deposit Account No. 02-2448 for any additional fees required under 37 C.F.R. §§ 1.16 or 1.17; particularly, extension of time fees.

Respectfully submitted,

BIRCH, STEWART, KOLASCH &, BIRCH, LLP

By: _____



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IFW

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PTO/SB/17 (07-06)

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U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE

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Effective on 12/08/2004.
Fees pursuant to the Consolidated Appropriations Act, 2005 (H.R. 4818).

FEE TRANSMITTAL For FY 2006

☐ Applicant claims small entity status. See 37 CFR 1.27

TOTAL AMOUNT OF PAYMENT (\$)
450.00

Complete if Known

Application Number	09/830,114-Conf. #3008
Filing Date	June 20, 2001
First Named Inventor	Bruno BESSETTE
Examiner Name	A. A. Armstrong
Art Unit	2626
Attorney Docket No.	4510-0105P

METHOD OF PAYMENT (check all that apply)

☒ Check ☐ Credit Card ☐ Money Order ☐ None ☐ Other (please identify): _____

☐ Deposit Account Deposit Account Number: 02-2448 Deposit Account Name: Birch, Stewart, Kolasch & Birch, LLP

For the above-identified deposit account, the Director is hereby authorized to: (check all that apply)

☐ Charge fee(s) indicated below ☐ Charge fee(s) indicated below, **except for the filing fee**

☒ Charge any additional fee(s) or underpayments of fee(s) under 37 CFR 1.16 and 1.17 ☒ Credit any overpayments

FEE CALCULATION

1. BASIC FILING, SEARCH, AND EXAMINATION FEES

Application Type	FILING FEES		SEARCH FEES		EXAMINATION FEES		Fees Paid (\$)
	Fee (\$)	Small Entity Fee (\$)	Fee (\$)	Small Entity Fee (\$)	Fee (\$)	Small Entity Fee (\$)	
Utility	300	150	500	250	200	100	
Design	200	100	100	50	130	65	
Plant	200	100	300	150	160	80	
Reissue	300	150	500	250	600	300	
Provisional	200	100	0	0	0	0	

2. EXCESS CLAIM FEES

Fee Description	Fee (\$)	Small Entity Fee (\$)
Each claim over 20 (including Reissues)	50	25
Each independent claim over 3 (including Reissues)	200	100
Multiple dependent claims	360	180

<u>Total Claims</u>	<u>Extra Claims</u>	<u>Fee (\$)</u>	<u>Fee Paid (\$)</u>	<u>Multiple Dependent Claims</u>
63	- 63 = 0	x 50.00 =	0.00	<u>Fee (\$)</u> <u>Fee Paid (\$)</u>

HP = highest number of total claims paid for, if greater than 20.

<u>Indep. Claims</u>	<u>Extra Claims</u>	<u>Fee (\$)</u>	<u>Fee Paid (\$)</u>
2	- 3 = 0	x 200.00 =	0.00

HP = highest number of independent claims paid for, if greater than 3.

3. APPLICATION SIZE FEE

If the specification and drawings exceed 100 sheets of paper (excluding electronically filed sequence or computer listings under 37 CFR 1.52(e)), the application size fee due is \$250 (\$125 for small entity) for each additional 50 sheets or fraction thereof. See 35 U.S.C. 41(a)(1)(G) and 37 CFR 1.16(s).

<u>Total Sheets</u>	<u>Extra Sheets</u>	<u>Number of each additional 50 or fraction thereof</u>	<u>Fee (\$)</u>	<u>Fee Paid (\$)</u>
	- 100 =	/50	(round up to a whole number) x	

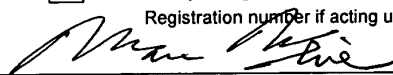
4. OTHER FEE(S)

Non-English Specification, \$130 fee (no small entity discount)

Other (e.g., late filing surcharge): 1252 Extension for response within second month 450.00

SUBMITTED BY			
Signature		Registration No. (Attorney/Agent)	32,181
Name (Print/Type)	Marc S. Weiner	Telephone	(703) 205-8000
		Date	November 8, 2006



PETITION FOR EXTENSION OF TIME UNDER 37 CFR 1.136(a) FY 2006 (Fees pursuant to the Consolidated Appropriations Act, 2005 (H.R. 4818).)		PTO/SB/22 (09-06) Approved for use through 03/31/2007. OMB 0651-0031 U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE 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.	
Application Number 09/830,114-Conf. #3008		Docket Number (Optional) 4510-0105P	
Filed June 20, 2001			
For A METHOD AND DEVICE FOR ADAPTIVE BANDWIDTH PITCH SEARCH IN CODING WIDEBAND SIGNALS			
Art Unit 2626		Examiner A. A. Armstrong	
This is a request under the provisions of 37 CFR 1.136(a) to extend the period for filing a reply in the above identified application. The requested extension and fee are as follows (check time period desired and enter the appropriate fee below):			
		<u>Fee</u>	<u>Small Entity Fee</u>
☐	One month (37 CFR 1.17(a)(1))	\$120	\$60 \$
☒	Two months (37 CFR 1.17(a)(2))	\$450	\$225 \$ 450.00
☐	Three months (37 CFR 1.17(a)(3))	\$1020	\$510 \$
☐	Four months (37 CFR 1.17(a)(4))	\$1590	\$795 \$
☐	Five months (37 CFR 1.17(a)(5))	\$2160	\$1080 \$
☐	Applicant claims small entity status. See 37 CFR 1.27.		
☒	A check in the amount of the fee is enclosed.		
☐	Payment by credit card. Form PTO-2038 is attached.		
☐	The Director has already been authorized to charge fees in this application to a Deposit Account.		
☒	The Director is hereby authorized to charge any fees which may be required, or credit any overpayment, to Deposit Account Number 02-2448. I have enclosed a duplicate copy of this sheet.		
I am the	☐	applicant/inventor.	
	☐	assignee of record of the entire interest. See 37 CFR 3.71. Statement under 37 CFR 3.73(b) is enclosed. (Form PTO/SB/96).	
	☒	attorney or agent of record. Registration Number 32,181	
	☐	attorney or agent under 37 CFR 1.34. Registration number if acting under 37 CFR 1.34	
		November 8, 2006	
Signature		Date	
Marc S. Weiner		(703) 205-8000	
Typed or printed name		Telephone Number	
NOTE: Signatures of all the inventors or assignees of record of the entire interest or their representative(s) are required. Submit multiple forms if more than one signature is required, see below.			
☐	Total of 1 forms are submitted.		

11/09/2006 SZEWDIE1 00000084 09830114

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PATENT APPLICATION FEE DETERMINATION RECORD Substitute for Form PTO-875					Application or Docket Number 09/830,114		Filing Date 06/20/2001		☐ To be Mailed			
APPLICATION AS FILED – PART I												
(Column 1)			(Column 2)			SMALL ENTITY ☐ OR		OTHER THAN SMALL ENTITY				
FOR		NUMBER FILED	NUMBER EXTRA		RATE (\$)	FEE (\$)		RATE (\$)	FEE (\$)			
☐ BASIC FEE (37 CFR 1.16(a), (b), or (c))		N/A	N/A		N/A			N/A				
☐ SEARCH FEE (37 CFR 1.16(k), (i), or (m))		N/A	N/A		N/A			N/A				
☐ EXAMINATION FEE (37 CFR 1.16(o), (p), or (q))		N/A	N/A		N/A			N/A				
TOTAL CLAIMS (37 CFR 1.16(i))		minus 20 =	*		X \$ =		OR	X \$ =				
INDEPENDENT CLAIMS (37 CFR 1.16(h))		minus 3 =	*		X \$ =			X \$ =				
☐ APPLICATION SIZE FEE (37 CFR 1.16(s))		If the specification and drawings exceed 100 sheets of paper, the application size fee due is \$250 (\$125 for small entity) for each additional 50 sheets or fraction thereof. See 35 U.S.C. 41(a)(1)(G) and 37 CFR 1.16(s).										
☐ MULTIPLE DEPENDENT CLAIM PRESENT (37 CFR 1.16(j))												
* If the difference in column 1 is less than zero, enter "0" in column 2.												
APPLICATION AS AMENDED – PART II										TOTAL		
(Column 1)			(Column 2)			(Column 3)			SMALL ENTITY OR OTHER THAN SMALL ENTITY			
AMENDMENT	11/08/2006	CLAIMS REMAINING AFTER AMENDMENT		HIGHEST NUMBER PREVIOUSLY PAID FOR	PRESENT EXTRA	RATE (\$)	ADDITIONAL FEE (\$)		RATE (\$)	ADDITIONAL FEE (\$)		
	Total (37 CFR 1.16(o))	* 63	Minus	** 63	= 0	X \$ =		OR	X \$50=	0		
	Independent (37 CFR 1.16(h))	* 2	Minus	***3	= 0	X \$ =		OR	X \$200=	0		
	☐ Application Size Fee (37 CFR 1.16(s))											
	☐ FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))											
	TOTAL ADD'L FEE										OR	TOTAL ADD'L FEE
(Column 1)			(Column 2)			(Column 3)						
AMENDMENT		CLAIMS REMAINING AFTER AMENDMENT		HIGHEST NUMBER PREVIOUSLY PAID FOR	PRESENT EXTRA	RATE (\$)	ADDITIONAL FEE (\$)		RATE (\$)	ADDITIONAL FEE (\$)		
	Total (37 CFR 1.16(o))	*	Minus	**	=	X \$ =		OR	X \$ =			
	Independent (37 CFR 1.16(h))	*	Minus	***	=	X \$ =		OR	X \$ =			
	☐ Application Size Fee (37 CFR 1.16(s))											
	☐ FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))											
	TOTAL ADD'L FEE										OR	TOTAL ADD'L 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.												

Legal Instrument Examiner:
margaret byars

This collection of information is required by 37 CFR 1.16. 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 1.14. 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.**

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EAST Search History

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S4	331	pitch same codebook same search	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/02/03 17:22

EAST Search History

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L8	0	("2005/0108005").URPN.	USPAT	OR	ON	2007/02/03 16:57
L7	92	3 with filter	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/02/03 16:57
L6	15	("4932061" "5307441" "5359696" "5444816" "5519807" "5664055" "5699482" "5701392" "5754976" "5963898" "6006174" "6064962" "6192334" "6449590").PN. OR ("6807524"). URPN.	US-PGPUB; USPAT; USOCR	OR	ON	2007/02/03 16:41

EAST Search History

L5	13	("5113262" "5235669" "5394473" "5444816" "5455888" "5581652" "5699482" "5701392" "5754976" "5956624" "5978759" "5999897" "6134373").PN. OR ("7151802").URPN.	US-PGPUB; USPAT; USOCR	OR	ON	2007/02/03 16:35
L4	32	1 and 3	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/02/03 16:31
L2	453	pitch adj3 search	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/02/03 16:30
L1	916	wideband with encod\$	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/02/03 16:30
S68	13	("5113262" "5235669" "5394473" "5444816" "5455888" "5581652" "5699482" "5701392" "5754976" "5956624" "5978759" "5999897" "6134373").PN. OR ("7151802").URPN.	US-PGPUB; USPAT; USOCR	OR	ON	2007/02/01 10:18
S67	15	("4932061" "5307441" "5359696" "5444816" "5519807" "5664055" "5699482" "5701392" "5754976" "5963898" "6006174" "6064962" "6192334" "6449590").PN. OR ("6807524").URPN.	US-PGPUB; USPAT; USOCR	OR	ON	2007/01/27 18:51
S66	100	("5873060" "6141638" "6795805" "20050108005" "20060277036" "5434947" "5570453" "5675702" "5692101" "5826224" "5581652" "6807524" "20050065785" "20050108007" "5457783" "5864797" "5396576" "5487128" "5659661" "5666464" "5717824" "5724480" "5774840" "5778334" "5963896" "6009388" "6023672" "6052660" "6073092" "6173257" "6393391" "6751585" "6980951" "7047188" "20020029140" "20020069052" "20020072904" "20040093205" "20040172402" "20040176950" "20050137863" "5826226" "20040073420" "20050177364" "20060106600" "20040111257" "5754976" "4963034" "5704000" "6236961" "5455888" "5924062" "6691083" "6704711" "20010044722" "20050154584" "20050165603" "20050251387" "20050267746" "20060100859" "5699485" "5884253" "5903866" "5924061" "4860355" "5664055" "5732389" "6014618" "4791670" "4969192" "5664054" "4980916" "5359696" "5528723" "5633982" "5657419" "5787389" "5793930" "5828811" "5839098" "5875423" "5913187" "6006178" "6014622" "6052659" "6055496" "6098037" "6199035" "6249758" "5199076" "5233660" "5245662" "5293449" "5295224" "5305332" "5307441" "5307460" "5371853" "5426718" "5491771").pn.	US-PGPUB; USPAT	OR	ON	2007/01/26 17:04
S65	97	S62 same S64	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/01/26 17:01

EAST Search History

S64	440	pitch with prediction with (error residual)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/01/26 17:01
S37	89	S34 same S36	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/01/26 17:01
S63	227	S61 same S62	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/01/26 17:00
S62	1155	(pitch adaptive) with codebook	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/01/26 17:00
S61	5026	pitch with analy\$4	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/01/26 17:00
S35	209	S33 same S34	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/01/26 17:00
S60	294	S59 and "704"/\$.ccls.	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/01/26 16:58
S59	831	S57 same S58	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/01/26 16:58
S58	3049	long with term with predict\$4	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/01/26 16:58
S57	1793	short with term with predict\$4	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/01/26 16:58
S43	275	S42 and "704"/\$.ccls.	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/01/26 16:58
S56	13	S54 same S55	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/01/26 16:57

EAST Search History

S55	1123	weight\$3 with synthesi\$ with filter	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/01/26 16:56
S13	968	weight\$ with synthesi\$ with filter	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/01/26 16:56
S54	55	convol\$5 with pitch with \$4vector	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/01/26 16:55
S14	8	S12 same S13	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/01/26 16:55
S53	15	("4932061" "5307441" "5359696" "5444816" "5519807" "5664055" "5699482" "5701392" "5754976" "5963898" "6006174" "6064962" "6192334" "6449590").PN. OR ("6807524"). URPN.	US-PGPUB; USPAT; USOCR	OR	ON	2007/01/26 16:52
S52	13	("5113262" "5235669" "5394473" "5444816" "5455888" "5581652" "5699482" "5701392" "5754976" "5956624" "5978759" "5999897" "6134373").PN. OR ("7151802").URPN.	US-PGPUB; USPAT; USOCR	OR	ON	2007/01/26 16:51
S51	0	("2005/0065785").URPN.	USPAT	OR	ON	2007/01/26 16:50
S50	0	(10/965795).APP.	USPAT; USOCR	OR	ON	2007/01/26 16:50
S49	0	("2005/0108007").URPN.	USPAT	OR	ON	2007/01/26 16:50
S48	65	S46 and S47	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/01/26 16:48
S47	398	pitch same codebook same search	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/01/26 16:48
S46	225	pitch with prediction with error	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/01/26 16:48
S5	54	S3 and S4	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	OR	ON	2007/01/26 16:48



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NOTICE OF ALLOWANCE AND FEE(S) DUE

2292 7590 02/08/2007

BIRCH STEWART KOLASCH & BIRCH
PO BOX 747
FALLS CHURCH, VA 22040-0747

EXAMINER

ARMSTRONG, ANGELA A

ART UNIT

PAPER NUMBER

2626

DATE MAILED: 02/08/2007

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.

09/830,114

06/20/2001

Bruno Bessette

4082-0129P

3008

TITLE OF INVENTION: METHOD AND DEVICE FOR ADAPTIVE BANDWIDTH PITCH SEARCH IN CODING WIDEBAND SIGNALS

APPLN. TYPE	SMALL ENTITY	ISSUE FEE DUE	PUBLICATION FEE DUE	PREV. PAID ISSUE FEE	TOTAL FEE(S) DUE	DATE DUE
nonprovisional	NO	\$1400	\$0	\$0	\$1400	05/08/2007

THE APPLICATION IDENTIFIED ABOVE HAS BEEN EXAMINED AND IS ALLOWED FOR ISSUANCE AS A PATENT. PROSECUTION ON THE MERITS IS CLOSED. THIS NOTICE OF ALLOWANCE 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.

THE ISSUE FEE AND PUBLICATION FEE (IF REQUIRED) 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. SEE 35 U.S.C. 151. THE ISSUE FEE DUE INDICATED ABOVE DOES NOT REFLECT A CREDIT FOR ANY PREVIOUSLY PAID ISSUE FEE IN THIS APPLICATION. IF AN ISSUE FEE HAS PREVIOUSLY BEEN PAID IN THIS APPLICATION (AS SHOWN ABOVE), THE RETURN OF PART B OF THIS FORM WILL BE CONSIDERED A REQUEST TO REAPPLY THE PREVIOUSLY PAID ISSUE FEE TOWARD THE ISSUE FEE NOW DUE.

HOW TO REPLY 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 the same, pay the TOTAL FEE(S) DUE shown above.

B. If the status above is to be removed, check box 5b on Part B - Fee(s) Transmittal and pay the PUBLICATION FEE (if required) and twice the amount of the ISSUE FEE shown above, or

If the SMALL ENTITY is shown as NO:

A. Pay TOTAL FEE(S) DUE shown above, or

B. If applicant claimed SMALL ENTITY status before, or is now claiming SMALL ENTITY status, check box 5a on Part B - Fee(s) Transmittal and pay the PUBLICATION FEE (if required) and 1/2 the ISSUE FEE shown above.

II. PART B - FEE(S) TRANSMITTAL, or its equivalent, must be completed and returned to the United States Patent and Trademark Office (USPTO) with your ISSUE FEE and PUBLICATION FEE (if required). If you are charging the fee(s) to your deposit account, section "4b" of Part B - Fee(s) Transmittal should be completed and an extra copy of the form should be submitted. If an equivalent of Part B is filed, a request to reapply a previously paid issue fee must be clearly made, and delays in processing may occur due to the difficulty in recognizing the paper as an equivalent of Part B.

III. All communications regarding this application must give the application number. Please direct all communications prior to issuance to Mail Stop 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.

PART B - FEE(S) TRANSMITTAL

**Complete and send this form, together with applicable fee(s), to: Mail Mail Stop ISSUE FEE
Commissioner for Patents
P.O. Box 1450
Alexandria, Virginia 22313-1450
or Fax (571)-273-2885**

INSTRUCTIONS: This form should be used for transmitting the ISSUE FEE and PUBLICATION FEE (if required). Blocks 1 through 5 should be completed where appropriate. All further correspondence including 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: Use Block 1 for any change of address)

Note: A certificate of mailing can only be used for domestic mailings of the Fee(s) 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 or transmission.

2292 7590 02/08/2007

**BIRCH STEWART KOLASCH & BIRCH
PO BOX 747
FALLS CHURCH, VA 22040-0747**

Certificate of Mailing or Transmission

I hereby certify that this Fee(s) Transmittal is being deposited with the United States Postal Service with sufficient postage for first class mail in an envelope addressed to the Mail Stop ISSUE FEE address above, or being facsimile transmitted to the USPTO (571) 273-2885, on the date indicated below.

(Depositor's name)
(Signature)
(Date)

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.

09/830,114 06/20/2001 Bruno Bessette 4082-0129P 3008

TITLE OF INVENTION: METHOD AND DEVICE FOR ADAPTIVE BANDWIDTH PITCH SEARCH IN CODING WIDEBAND SIGNALS

APPLN. TYPE	SMALL ENTITY	ISSUE FEE DUE	PUBLICATION FEE DUE	PREV. PAID ISSUE FEE	TOTAL FEE(S) DUE	DATE DUE

nonprovisional NO \$1400 \$0 \$0 \$1400 05/08/2007

EXAMINER	ART UNIT	CLASS-SUBCLASS

ARMSTRONG, ANGELA A 2626 704-207000

1. Change of correspondence address or indication of "Fee Address" (37 CFR 1.363).

- ☐ 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; Rev 03-02 or more recent) attached. **Use of a Customer Number is required.**

2. For printing on the patent front page, list

- (1) the names of up to 3 registered patent attorneys or agents OR, alternatively, 1 _____
(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. 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. If an assignee is identified below, the document has been filed for recordation as set forth in 37 CFR 3.11. Completion of this form is NOT a substitute for filing an assignment.

(A) NAME OF ASSIGNEE

(B) RESIDENCE: (CITY and STATE OR COUNTRY)

Please check the appropriate assignee category or categories (will not be printed on the patent): ☐ Individual ☐ Corporation or other private group entity ☐ Government

4a. The following fee(s) are submitted:

- ☐ Issue Fee
☐ Publication Fee (No small entity discount permitted)
☐ Advance Order - # of Copies _____

4b. Payment of Fee(s): (Please first reapply any previously paid issue fee shown above)

- ☐ A check is enclosed.
☐ Payment by credit card. Form PTO-2038 is attached.
☐ The Director is hereby authorized to charge the required fee(s), any deficiency, or credit any overpayment, to Deposit Account Number _____ (enclose an extra copy of this form).

5. Change in Entity Status (from status indicated above)

- ☐ a. Applicant claims SMALL ENTITY status. See 37 CFR 1.27. ☐ b. Applicant is no longer claiming SMALL ENTITY status. See 37 CFR 1.27(g)(2).

NOTE: The Issue Fee and Publication Fee (if required) 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 United States Patent and Trademark Office.

Authorized Signature _____

Date _____

Typed or printed name _____

Registration No. _____

This collection of information is required by 37 CFR 1.311. 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 1.14. 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, Virginia 22313-1450. **DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, Virginia 22313-1450.**

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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/830,114	06/20/2001	Bruno Bessette	4082-0129P	3008
2292	7590	02/08/2007		
BIRCH STEWART KOLASCH & BIRCH PO BOX 747 FALLS CHURCH, VA 22040-0747				
			EXAMINER ARMSTRONG, ANGELA A	
			ART UNIT 2626	PAPER NUMBER
DATE MAILED: 02/08/2007				

Determination of Patent Term Extension under 35 U.S.C. 154 (b) (application filed after June 7, 1995 but prior to May 29, 2000)

The Patent Term Extension is 0 day(s). Any patent to issue from the above-identified application will include an indication of the 0 day extension on the front page.

If a Continued Prosecution Application (CPA) was filed in the above-identified application, the filing date that determines Patent Term Extension is the filing date of the most recent CPA.

Applicant will be able to obtain more detailed information by accessing the Patent Application Information Retrieval (PAIR) WEB site (<http://pair.uspto.gov>).

Any questions regarding the Patent Term Extension or Adjustment determination should be directed to the Office of Patent Legal Administration at (571)-272-7702. Questions relating to issue and publication fee payments should be directed to the Customer Service Center of the Office of Patent Publication at 1-(888)-786-0101 or (571)-272-4200.

Notice of Allowability	Application No.	Applicant(s)	
	09/830,114	BESSETTE ET AL.	
	Examiner	Art Unit	
	Angela A. Armstrong	2626	

-- 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 (PTOL-85) 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 the amendment filed 11/08/06.

2. ☒ The allowed claim(s) is/are 1-63.

3. ☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).
 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: _____.

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.

4. ☐ A SUBSTITUTE OATH OR DECLARATION must be submitted. Note the attached EXAMINER'S AMENDMENT or NOTICE OF INFORMAL PATENT APPLICATION (PTO-152) which gives reason(s) why the oath or declaration is deficient.

5. ☐ CORRECTED DRAWINGS (as "replacement sheets") must be submitted.
 (a) ☐ including changes required by the Notice of Draftsperson's Patent Drawing Review (PTO-948) attached
 1) ☐ hereto or 2) ☐ to Paper No./Mail Date _____.
 (b) ☐ including changes required by the attached Examiner's Amendment / Comment or in the Office action of Paper No./Mail Date _____.

Identifying indicia such as the application number (see 37 CFR 1.84(c)) should be written on the drawings in the front (not the back) of each sheet. Replacement sheet(s) should be labeled as such in the header according to 37 CFR 1.121(d).

6. ☐ DEPOSIT OF and/or INFORMATION about the deposit of BIOLOGICAL MATERIAL must be submitted. Note the attached Examiner's comment regarding REQUIREMENT FOR THE DEPOSIT OF BIOLOGICAL MATERIAL.

Attachment(s)

1. ☐ Notice of References Cited (PTO-892) 2. ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948) 3. ☐ Information Disclosure Statements (PTO/SB/08), Paper No./Mail Date _____ 4. ☐ Examiner's Comment Regarding Requirement for Deposit of Biological Material	5. ☐ Notice of Informal Patent Application 6. ☐ Interview Summary (PTO-413), Paper No./Mail Date _____ 7. ☐ Examiner's Amendment/Comment 8. ☐ Examiner's Statement of Reasons for Allowance 9. ☐ Other _____


Angela A. Armstrong
Primary Examiner
Art Unit: 2626

Search Notes 	Application/Control No. 09830114	Applicant(s)/Patent Under Reexamination BESSETTE ET AL.
	Examiner Armstrong, Angela A	Art Unit 2626

SEARCHED			
Class	Subclass	Date	Examiner

SEARCH NOTES		
Search Notes	Date	Examiner
PLUS search	1/23/07	AAA
EAST text search	1/26/07	AAA
EAST text search	1/27/07	AAA
EAST text search	02/01/07	AAA
EAST text search	02/03/07	AAA

INTERFERENCE SEARCH			
Class	Subclass	Date	Examiner
704	219,223	02/03/07	AAA

Index of Claims 	Application/Control No. 09830114	Applicant(s)/Patent Under Reexamination BESSETTE ET AL.
	Examiner Armstrong, Angela A	Art Unit 2626

✓	Rejected	-	Cancelled	N	Non-Elected	A	Appeal
=	Allowed	÷	Restricted	I	Interference	O	Objected

☐ Claims renumbered in the same order as presented by applicant			☐ CPA			☐ T.D.			☐ R.1.47		
CLAIM		DATE									
Final	Original	02/05/2007									
1	1	=									
2	2	=									
3	3	=									
4	4	=									
5	5	=									
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24	33	=									
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25	35	=									
27	36	=									

Index of Claims 	Application/Control No. 09830114	Applicant(s)/Patent Under Reexamination BESSETTE ET AL.
	Examiner Armstrong, Angela A	Art Unit 2626

✓	Rejected	-	Cancelled	N	Non-Elected	A	Appeal
=	Allowed	÷	Restricted	I	Interference	O	Objected

☐ Claims renumbered in the same order as presented by applicant			☐ CPA			☐ T.D.			☐ R.1.47			
CLAIM			DATE									
Final	Original		02/05/2007									
28	37	=										
29	38	=										
30	39	=										
31	40	=										
32	41	=										
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35	43	=										
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52	61	=										
53	62	=										
54	63	=										

09/830114

09/830114

JC18 Rec'd PCT/PTO 2 4 APR 2001

PATENT
4082-0129P

IN THE U.S. PATENT AND TRADEMARK OFFICE

Applicant: Bruno BESSETTE et al. Conf.:
Int'l. Appl. No.: PCT/CA99/01008
Appl. No.: NEW Group:
Filed: April 24, 2001 Examiner:
For: A METHOD AND DEVICE FOR ADAPTIVE
BANDWIDTH PITCH SEARCH IN CODING
WIDEBAND SIGNALS

PRELIMINARY AMENDMENT

BOX PATENT APPLICATION

Assistant Commissioner for Patents
Washington, DC 20231

April 24, 2001

Sir:

The following Preliminary Amendments and Remarks are respectfully submitted in connection with the above-identified application.

AMENDMENTS

IN THE SPECIFICATION:

Please ^{insert} amend the specification as follows:

Page 1, after the heading "BACKGROUND OF THE INVENTION", insert the paragraph --This application is the national phase under 35 U.S.C. § 371 of PCT International Application No. PCT/CA99/01008 which has an International filing date of October 27, 1999, which designated the United States of America and was published in English.



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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/830,114	06/20/2001	Bruno Bessette	4082-0129P	3008
2292	7590	02/28/2007		
BIRCH STEWART KOLASCH & BIRCH PO BOX 747 FALLS CHURCH, VA 22040-0747			EXAMINER ARMSTRONG, ANGELA A	
			ART UNIT 2626	PAPER NUMBER

DATE MAILED: 02/28/2007

PRIORITY ACKNOWLEDGMENT

- ☒ 1. Receipt is acknowledged of priority papers submitted under 35 U.S.C. 119. The papers have been placed of record in the file.
- ☐ 2. Applicant's claim for priority, based on papers filed in parent Application Number _____ submitted under 35 U.S.C. 119, is acknowledged.
- ☐ 3. The priority papers, submitted _____, after payment of the issue fee are
- ☐ acknowledged
While the priority claim or certified copy filed will be placed in the file record, neither will be reviewed and the patent when published will not include the priority claim.
See 37 CFR 1.55(a)(2).
 - ☐ not acknowledged since the processing fee in 37 CFR 1.17(i) has not been received.
- ☐ 4. For utility and plant applications filed on or after November 29, 2000, the priority claim is not entered because the claim was not presented within the time limit required by 37 CFR 1.55(a)(1). A petition to accept a delayed claim for priority under 35 U.S.C. 119(a) - (d) or (f), or 365(a) may be filed. See 37 CFR 1.55(c) and MPEP 201.14(a).

James M. Schell, Jr.
Manager, Publishing Division
Office of Patent Publication
(703) 305-8388

PART B - FEE(S) TRANSMITTAL

Complete and send this form, together with applicable fee(s), to: **Mail** Mail Stop ISSUE FEE
 Commissioner for Patents
 P.O. Box 1450
 Alexandria, Virginia 22313-1450
 or **Fax** (571)-273-2885

INSTRUCTIONS: This form should be used for transmitting the ISSUE FEE and PUBLICATION FEE (if required). Blocks 1 through 5 should be completed where appropriate. All further correspondence including 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: Use Block 1 for any change of address)

2292 7590 02/08/2007

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Note: A certificate of mailing can only be used for domestic mailings of the Fee(s) 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 or transmission.

Certificate of Mailing or Transmittal
 I hereby certify that this Fee(s) Transmittal is being deposited with the United States Postal Service with sufficient postage for first class mail in an envelope addressed to the Mail Stop ISSUE FEE address above, or being facsimile transmitted to the USPTO (571) 273-2885, on the date indicated below.

(Depositor's name)
(Signature)
(Date)

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/830,114	06/20/2001	Bruno Bessette	4882-0129P 4510-0108	3008

TITLE OF INVENTION: METHOD AND DEVICE FOR ADAPTIVE BANDWIDTH FITCH SEARCH IN CODING WIDEBAND SIGNALS

APPLN. TYPE	SMALL ENTITY	ISSUE FEE DUE	PUBLICATION FEE DUE	PREV. PAID ISSUE FEE	TOTAL FEE(S) DUE	DATE DUE
nonprovisional	NO	\$1400	\$0	\$0	\$1400	05/08/2007

EXAMINER	ART UNIT	CLASS-SUBCLASS
ARMSTRONG, ANGELA A	2626	704-207000

1. Change of correspondence address or indication of "Fee Address" (37 CFR 1.363).
☐ 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; Rev 03-02 or more recent) attached. Use of a Customer Number is required.

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.

BIRCH STEWART KOLASCH
 & BIRCH, LLP
 1
 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. If an assignee is identified below, the document has been filed for recordation as set forth in 37 CFR 3.11. Completion of this form is NOT a substitute for filing an assignment.

(A) NAME OF ASSIGNEE

VOICEAGE CORPORATION

(B) RESIDENCE: (CITY and STATE OR COUNTRY)

Quebec, Canada

Please check the appropriate assignee category or categories (will not be printed on the patent): ☐ Individual ☒ Corporation or other private group entity ☐ Government

4a. The following fee(s) are submitted:

☒ Issue Fee
☐ Publication Fee (No small entity discount permitted)
☒ Advance Order - # of Copies 4 (four)

4b. Payment of Fee(s): (Please first reapply any previously paid issue fee shown above)

☐ A check is enclosed.
☐ Payment by credit card. Form PTO-2038 is attached. (if necessary)
☒ The Director is hereby authorized to charge the required fee(s), any deficiency, or credit any overpayment, to Deposit Account Number 02 2448 (enclose an extra copy of this form).

5. Change in Entity Status (from status indicated above)

☐ a. Applicant claims SMALL ENTITY status. See 37 CFR 1.27. ☐ b. Applicant is no longer claiming SMALL ENTITY status. See 37 CFR 1.27(b)(2).

NOTE: The Issue Fee and Publication Fee (if required) 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 United States Patent and Trademark Office.

Authorized Signature

Marc S. Weiner #32181

Date

May 8, 2007

Typed or printed name

Marc S. Weiner

Registration No. 32,181

This collection of information is required by 37 CFR 1.311. 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 1.14. 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, Virginia 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, Virginia 22313-1450.

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PTOL-85 (Rev. 07/06) Approved for use through 04/	05/09/2007	00000320	1	1501	\$1,400.00	05/08/2007
PAGE 2/2 * RCVD AT 7/18/2007 5:37:25 PM [Eastern Da	05/09/2007	00000321	1	8001	\$12.00	05/08/2007



UNITED STATES PATENT AND TRADEMARK OFFICE

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APPLICATION NO.	ISSUE DATE	PATENT NO.	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/830,114	08/21/2007	7260521	4082-0129P	3008

2292 7590 08/01/2007
BIRCH STEWART KOLASCH & BIRCH
PO BOX 747
FALLS CHURCH, VA 22040-0747

ISSUE NOTIFICATION

The projected patent number and issue date are specified above.

Determination of Patent Term Extension under 35 U.S.C. 154 (b) (application filed after June 7, 1995 but prior to May 29, 2000)

The Patent Term Extension is 0 day(s). Any patent to issue from the above-identified application will include an indication of the 0 day extension on the front page.

If a Continued Prosecution Application (CPA) was filed in the above-identified application, the filing date that determines Patent Term Extension is the filing date of the most recent CPA.

Any questions regarding the Patent Term Extension or Adjustment determination should be directed to the Office of Patent Legal Administration at (571)-272-7702. Questions relating to issue and publication fee payments should be directed to the Customer Service Center of the Office of Patent Publication at (571)-272-4200.

APPLICANT(s) (Please see PAIR WEB site <http://pair.uspto.gov> for additional applicants):

Bruno Bessette, Quebec, CANADA;
Redwan Salami, Quebec, CANADA;
Roch Lefebvre, Magog, CANADA;

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JUL 03 2008

TELECOPIER TRANSMISSION

Date: July 3, 2008
Number of pages including this one: - 3 -

TO:
Company: USPTO
Fax: 1-571-273-8300

FROM: Name: Gaetan Prince
Direct line: (514)397-6725
E-mail: gp@bcf.ca
Ref. No.: 08241-078

Operator: Todd LeSage
Telephone: (514)397-8500
Extension: 3448

COMMENTS:

DUE DATE: AS SOON AS POSSIBLE

U.S. PATENT NO.: 7,260,521

ATTORNEY DOCKET NO.: 08241-078

ENCLOSURES:

1. Letter to the Commissioner of Patents;
2. Form PTO/SB/47.

sent on July 3, 2008

STAMP, DATE AND RETURN

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JUL 03 2008**

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

U.S. Patent No. : 7,260,521
Issue Date : August 21, 2007
Applicant/Owner : VOICEAGE CORPORATION
Title : A METHOD AND DEVICE FOR
ADAPTIVE BANDWIDTH PITCH SEARCH
IN CODING WIDEBAND SIGNALS
Attorney Docket No. : 08241-078

Montreal, Quebec, Canada
July 3, 2008

CHANGE OF CORRESPONDENCE ADDRESS

COMMISSIONER FOR PATENTS
United States Patent and Trademark Office
Customer Service Window, **Mail Stop Post Issue**
Randolph Building
401 Dulany Street
Alexandria, VA 22314

SIR:

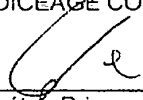
Please find enclosed a "Fee Address" Indication Form (Form PTO/SB/47) which has been duly executed by Mr. Sylvain Desjardins, the Executive Vice-President of VoiceAge Corporation.

Accordingly, please direct all further correspondence for the above-captioned patent to the undersigned the new address indicated below.

Finally, please note our new Docket Number for this case: 08241-078.

Respectfully submitted,

VOICEAGE CORPORATION



Gaétan Prince
Patent Agent Registration No. 33,107
BCF LLP
1100 René-Lévesque Blvd: West
25th Floor
Montreal, Quebec, Canada H3B 5C9
Tel: 514-397-6725
Fax: 514-397-8515

TO: Mail Stop 8 Director of the U.S. Patent and Trademark Office P.O. Box 1450 Alexandria, VA 22313-1450	REPORT ON THE FILING OR DETERMINATION OF AN ACTION REGARDING A PATENT OR TRADEMARK

In Compliance with 35 U.S.C. § 290 and/or 15 U.S.C. § 1116 you are hereby advised that a court action has been
filed in the U.S. District Court for the Eastern District of Texas on the following

☐ Trademarks or ☒ Patents. (☐ the patent action involves 35 U.S.C. § 292.);

DOCKET NO. 2:14-cv-1055	DATE FILED 11/18/2014	U.S. DISTRICT COURT for the Eastern District of Texas
PLAINTIFF SAINT LAWRENCE COMMUNICATIONS LLC		DEFENDANT LG ELECTRONICS, INC., LG ELECTRONICS ALABAMA, INC. AND LG ELECTRONICS USA, INC.
PATENT OR TRADEMARK NO.	DATE OF PATENT OR TRADEMARK	HOLDER OF PATENT OR TRADEMARK
1 6,795,805	9/21/2004	St. Lawrence
2 6,807,524	10/19/2004	St. Lawrence
3 7,151,802	12/19/2006	St. Lawrence
4 7,260,521	8/21/2007	St. Lawrence
5 7,191,123	3/13/2007	St. Lawrence

In the above—entitled case, the following patent(s)/ trademark(s) have been included:

DATE INCLUDED	INCLUDED BY ☐ Amendment ☐ Answer ☐ Cross Bill ☐ Other Pleading	
PATENT OR TRADEMARK NO.	DATE OF PATENT OR TRADEMARK	HOLDER OF PATENT OR TRADEMARK
1		
2		
3		
4		
5		

In the above—entitled case, the following decision has been rendered or judgement issued:

DECISION/JUDGEMENT		
CLERK	(BY) DEPUTY CLERK	DATE

Copy 1—Upon initiation of action, mail this copy to Director Copy 3—Upon termination of action, mail this copy to Director
Copy 2—Upon filing document adding patent(s), mail this copy to Director Copy 4—Case file copy

AO 120 (Rev. 08/10)

TO: Mail Stop 8 Director of the U.S. Patent and Trademark Office P.O. Box 1450 Alexandria, VA 22313-1450	REPORT ON THE FILING OR DETERMINATION OF AN ACTION REGARDING A PATENT OR TRADEMARK

In Compliance with 35 U.S.C. § 290 and/or 15 U.S.C. § 1116 you are hereby advised that a court action has been filed in the U.S. District Court Central District of California on the following

☐ Trademarks or ☒ Patents. (☐ the patent action involves 35 U.S.C. § 292.):

DOCKET NO.	DATE FILED 03/09/2015	U.S. DISTRICT COURT Central District of California
PLAINTIFF HTC Corporation, HTC America, Inc.		DEFENDANT Acacia Research Corporation, Saint Lawrence Communications LLC
PATENT OR TRADEMARK NO.	DATE OF PATENT OR TRADEMARK	HOLDER OF PATENT OR TRADEMARK
1 6,795,805	09/21/2004	Saint Lawrence Communications LLC
2 6,807,524	10/19/2004	Saint Lawrence Communications LLC
3 7,151,802	12/19/2006	Saint Lawrence Communications LLC
4 7,260,521	08/21/2007	Saint Lawrence Communications LLC
5 7,191,123	03/13/2007	Saint Lawrence Communications LLC

In the above—entitled case, the following patent(s)/ trademark(s) have been included:

DATE INCLUDED	INCLUDED BY	
	☐ Amendment ☐ Answer ☐ Cross Bill ☐ Other Pleading	
PATENT OR TRADEMARK NO.	DATE OF PATENT OR TRADEMARK	HOLDER OF PATENT OR TRADEMARK
1		
2		
3		
4		
5		

In the above—entitled case, the following decision has been rendered or judgement issued:

DECISION/JUDGEMENT

CLERK	(BY) DEPUTY CLERK	DATE

Copy 1—Upon initiation of action, mail this copy to Director Copy 3—Upon termination of action, mail this copy to Director
 Copy 2—Upon filing document adding patent(s), mail this copy to Director Copy 4—Case file copy