08/764656

34 | 514R
Class | Subclass
ISSUE CLASSIFICATION

[redacted]

5689440

UTILITY SERIAL NUMBER	PATENT DATE	PATENT NUMBER
08/764656	NOV 18 1997	5689440

SERIAL NUMBER	FILING DATE	CLASS	SUBCLASS	GROUP ART UNIT	EXAMINER
08/764,656	12/11/96	364	514R	2414	D. Smith

APPLICANTS: CLIFFORD D. LEITCH, CORAL SPRINGS, FL; ROBERT J. SCHWENDEMAN, POMPANO BEACH, FL; KAZIMIERZ SIWIAK, CORAL SPRINGS, FL; WILLIAM J. KUZNICKI, CORAL SPRINGS, FL; SUNIL SATYAMURTI, DELRAY BEACH, FL.

CONTINUING DATA******************* SMITH 364/514R
VERIFIED THIS APPLN IS A CON OF 08/395,747 02/28/95 abandoned

ok DS 3/12/97

FOREIGN/PCT APPLICATIONS**********
VERIFIED

none DS 3/12/97

FOREIGN FILING LICENSE GRANTED 02/20/97

Foreign priority claimed / 35 USC 119 conditions met / Verified and Acknowledged	AS FILED	STATE OR COUNTRY	SHEETS DRWGS.	TOTAL CLAIMS	INDEP. CLAIMS	FILING FEE RECEIVED	ATTORNEY'S DOCKET NO.
☐ yes ☒ no / ☐ yes ☒ no / Examiner's Initials	→	FL	14	24	7	$1,178.00	PT00600UC01

ADDRESS:
MOTOROLA INC
IP LAW DEPARTMENT MS 96
1500 GATEWAY BLVD
BOYNTON BEACH FL 33426-8753

TITLE: VOICE COMPRESSION METHOD AND APPARATUS IN A COMMUNICATION SYSTEM

U.S. DEPT. OF COMM./PAT. & TM—PTO-436L (Rev.12-94)

PARTS OF APPLICATION FILED SEPARATELY

L. Wee
Applications Examiner

NOTICE OF ALLOWANCE MAILED
3/19/97

Demetra R. Smith
Assistant Examiner

CLAIMS ALLOWED

Total Claims	Print Claim
~~24~~ 20	~~12~~

ISSUE FEE

Amount Due	Date Paid
$1290-00	6-16-97

ELLIS B. RAMIREZ
PRIMARY EXAMINER
GROUP 2400
Primary Examiner

DRAWING

Sheets Drwg.	Figs. Drwg.	Print Fig.
14	25	1

ISSUE BATCH NUMBER X90

PREPARED FOR ISSUE 3/25/97

Label Area

WARNING: The information disclosed herein may be restricted. Unauthorized disclosure may be prohibited by the United States Code Title 35, Sections 122, 181 and 368. Possession outside the U.S. Patent & Trademark Office is restricted to authorized employees and contractors only.

Form PTO-436A (Rev. 8/92)

JR

ISSUE FEE ... ME

46

(FACE)

08/395747

Class	Subclass
364	514R

ISSUE CLASSIFICATION

[redacted]

25

UTILITY SERIAL NUMBER	PATENT DATE	PATENT NUMBER
08/395747		

SERIAL NUMBER	FILING DATE	CLASS	SUBCLASS	GROUP ART UNIT	EXAMINER
08/395,747	02/28/95	~~381~~ 364	514R	~~2308~~ 2414	Smith ~~E. RAMIREZ~~

APPLICANTS CLIFFORD D. LEITCH, CORAL SPRINGS, FL; ROBERT J. SCHWENDEMAN, POMPANO BEACH, FL; KAZIMIERZ SIWIAK, CORAL SPRINGS, FL; WILLIAM J. KUZNICKI, CORAL SPRINGS, FL; SUNIL SATYAMURTI, DELRAY BEACH, FL.

CONTINUING DATA********************
VERIFIED

none DS 9/13/96

FOREIGN/PCT APPLICATIONS**********
VERIFIED

none DS 9/13/96

FOREIGN FILING LICENSE GRANTED 03/20/95

Foreign priority claimed / 35 USC 119 conditions met		STATE OR COUNTRY	SHEETS DRWGS.	TOTAL CLAIMS	INDEP. CLAIMS	FILING FEE RECEIVED	ATTORNEY'S DOCKET NO.
☐ yes ☒ no / ☐ yes ☒ no — Verified and Acknowledged — Examiner's Initials	AS FILED	FL	14	24	7	$1,122.00	PT00600U

ADDRESS
JOHN H. MOORE
MOTOROLA INC
1500 GATEWAY BLVD MS 96
BOYNTON BEACH FL 33426-8292

TITLE
VOICE COMPRESSION METHOD AND APPARATUS IN A COMMUNICATION SYSTEM

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

PARTS OF APPLICATION FILED SEPARATELY

Applications Examiner

NOTICE OF ALLOWANCE MAILED

10/1/96

Demetra R. Smith
Assistant Examiner

EMANUEL T. VOELTZ
SUPERVISORY PATENT EXAMINER
GROUP 2400

Primary Examiner

CLAIMS ALLOWED

Total Claims	Print Claim
24	12

DRAWING

Sheets Drwg.	Figs. Drwg.	Print Fig.
14	25	1

ISSUE FEE

Amount Due	Date Paid
$1290.00	

ISSUE BATCH NUMBER F03

Label Area

PREPARED FOR ISSUE

WARNING: The information disclosed herein may be restricted. Unauthorized disclosure may be prohibited by the United States Code Title 35, Sections 122, 181 and 368. Possession outside the U.S. Patent & Trademark Office is restricted to authorized employees and contractors only.

Form PTO-436A
(Rev. 8/92)

(FACE)

08/395747

PATENT APPLICATION

08395747

[APPR]OVED FOR LICENSE

ALS MAR 13 95

CONTENTS

RECEIVED

APR 04 1995

GROUP 2300

Date Entered or Counted		Date Received or Mailed
	1. Application 14 pts. papers.	2/28/95
	2. Prior Art	2/28/95
	3. Prior Art	4/3/95
10-1-96	4. Notice of [illegible]	10/1/96
	5. ABANDONED	3-17-97
	6.	
	7.	
	8.	
	9.	
	10.	
	11.	
	12.	
	13.	
	14.	
	15.	
	16.	
	17.	
	18.	
	19.	
	20.	
	21.	
	22.	
	23.	
	24.	
	25.	
	26.	
	27.	
	28.	
	29.	
	30.	
	31.	
	32.	

08/764656

PATENT APPLICATION

08764656

APPROVED FOR LICENSE

INITIALS

CONTENTS

Date Entered or Counted			Date Received or Mailed
		Application papers.	
	6.	Pre Amdt A	12-11-96
	7.	IDS	12-11-96
3-17-97	8.	Notice of Allowance	3-19-97
	9.	Ltr re: form 1449	03-28-97
	10.	Supplemental	4-24-97
	11.	letter	4-4-97
	12.	Interview Summary	6-12-97
6-16-97	13.	Suppl. Ex. Amdt B	
	14.	PTO 6... NOV 1 8 1997	
	15.		

(FRONT)

STAPLE AREA

★ U.S. GOVERNMENT PRINTING OFFICE: 1991—302-420

PATENT NUMBER

APPLICATION SERIAL NUMBER

08/395747

APPLICANT'S NAME (PLEASE PRINT)

Leitch et al

IF REISSUE, ORIGINAL PATENT NUMBER

INTERNATIONAL CLASSIFICATION

H	0	4	B		7/00

ORIGINAL CLASSIFICATION

CLASS	SUBCLASS
364	514R

CROSS REFERENCE(S)

CLASS	SUBCLASS (ONE SUBCLASS PER BLOCK)			
370	109			
455	54.1	70		
381	29	34		

GROUP ART UNIT: 2414

ASSISTANT EXAMINER (PLEASE STAMP OR PRINT FULL NAME): Demetra R. Smith

PRIMARY EXAMINER (PLEASE STAMP OR PRINT FULL NAME): Emanuel Todd Voeltz

PTO 270 (REV. 5-91)

ISSUE CLASSIFICATION SLIP

U.S. DEPARTMENT OF COMMERCE PATENT AND TRADEMARK OFFICE

Staple Issue Slip Here

POSITION		ID NO.	DATE
CLASSIFIER		12	3/13/95
EXAMINER		351	3-20-95
TYPIST		17	3-20
VERIFIER		81	3/22
CORPS CORR.			
SPEC. HAND			
FILE MAINT.			
DRAFTING			

INDEX OF CLAIMS

Claim		Date							
Final	Original	9/15/96							
1	1	=							
2	2	=							
3	3	↓							
4	4	↓							
5	5	↓							
6	6	↓							
7	7	↓							
8	8	✓							
9	9	=							
10	10	↓							
11	11	✓							
12	12	=							
13	13	↓							
14	14	↓							
15	15	✓							
16	16	=							
17	17	✓							
18	18	=							
19	19	↓							
20	20	↓							
21	21	↓							
22	22	✓							
23	23	=							
24	24	=							
	25								
	26								
	27								
	28								
	29								
	30								
	31								
	32								
	33								
	34								
	35								
	36								
	37								
	38								
	39								
	40								
	41								
	42								
	43								
	44								
	45								
	46								
	47								
	48								
	49								
	50								

SYMBOLS

- ✓ .... Rejected
- = ... Allowed
- – (Through numberal) Canceled
- ÷ .... Restricted
- N .... Non-elected
- I .... Interference
- A .... Appeal
- O .... Objected

Claim		Date							
Final	Original								
	51								
	52								
	53								
	54								
	55								
	56								
	57								
	58								
	59								
	60								
	61								
	62								
	63								
	64								
	65								
	66								
	67								
	68								
	69								
	70								
	71								
	72								
	73								
	74								
	75								
	76								
	77								
	78								
	79								
	80								
	81								
	82								
	83								
	84								
	85								
	86								
	87								
	88								
	89								
	90								
	91								
	92								
	93								
	94								
	95								
	96								
	97								
	98								
	99								
	100								

☆ U.S. GOVERNMENT PRINTING OFFICE: 1996-410 219

POSITION		ID NO.	DATE
CLASSIFIER			
EXAMINER		405	1-13-97
TYPIST		442	2/20/97
VERIFIER		#570	2-21-97
CORPS CORR.			
SPEC. HAND			
FILE MAINT.			
DRAFTING			

INDEX OF CLAIMS

Claim Final	Claim Original	Date 3/12/97	Date 6/12/97	Date	Date	Date	Date	Date	Date
1	1	=	=						
.	~~2~~	=	[redacted]						
	~~3~~	=	[redacted]						
2	4	=	=						
3	5	=	=						
4	6	=	=						
5	7	=	=						
6	8	=	=						
7	9	=	=						
8	10	=	=						
9	11	=	=						
10	12	=	=						
11	13	=	=						
12	14	=	=						
	~~15~~	=	[redacted]						
13	16	=	=						
	~~17~~	=	[redacted]						
14	18	=	=						
15	19	=	=						
16	20	=	=						
17	21	=	=						
18	22	=	=						
19	23	=	=						
20	24	=	=						
	25								
	26								
	27								
	28								
	29								
	30								
	31								
	32								
	33								
	34								
	35								
	36								
	37								
	38								
	39								
	40								
	41								
	42								
	43								
	44								
	45								
	46								
	47								
	48								
	49								
	50								

SYMBOLS

✓ .... Rejected
= .... Allowed
– (Through numeral) Canceled
÷ .... Restricted
N .... Non-elected
I .... Interference
A .... Appeal
O .... Objected

Claim Final	Claim Original	Date	Date	Date	Date	Date	Date	Date	Date
	51								
	52								
	53								
	54								
	55								
	56								
	57								
	58								
	59								
	60								
	61								
	62								
	63								
	64								
	65								
	66								
	67								
	68								
	69								
	70								
	71								
	72								
	73								
	74								
	75								
	76								
	77								
	78								
	79								
	80								
	81								
	82								
	83								
	84								
	85								
	86								
	87								
	88								
	89								
	90								
	91								
	92								
	93								
	94								
	95								
	96								
	97								
	98								
	99								
	100								

(LEFT INSIDE)

[redacted]

SEARCHED

Class	Sub.	Date	Exmr.
381	29 30 35 36 37 34	9/6/96	DS
364 395	514R 2.14 2.12 2.21 2.24 2.29	9/6/96	DS
340	825.44		
455	54.1	9/8/96	DS
455	70	9/8/96	DS
455	72 109 47 38.1	9/10/96	DS

INTERFERENCE SEARCHED

Class	Sub.	Date	Exmr.
364	514R	10/4/96	DS
364	514R	3/12/97	DS

SEARCH NOTES

	Date	Exmr.
APS	9/5/96	DS
MAYA	9/6/96	DS
STIC-EIC Search	9/11/96	DS
Andrew Faile AU2611 -455-72, 109, 47 -455/72, 38.01	9/20/96	DS

(RIGHT OUTSIDE)

[redacted]

SEARCHED

Class	Sub.	Date	Exmr.
381	29	9/4/96	DS
	30		DS
	35		DS
	36		DS
	37		DS
	34		DS
			DS
364	514R	9/6/96	
395	2	9/6/96	DS
395	2.14	9/6/96	DS
	2.12	9/6/96	DS
	2.21	9/6/96	DS
	2.24	9/6/96	DS
	2.29	9/6/96	DS
340	825.44	9/6/96	DS
455	54.1	9/8/96	DS
455	70	9/8/96	DS
455	72	9/10/96	DS
455	109	9/10/96	DS
455	47	9/10/96	DS
~~[illegible]~~	~~[illegible]~~	~~[illegible]~~	~~[illegible]~~
455	38.1	9/10/96	DS

INTERFERENCE SEARCHED

Class	Sub.	Date	Exmr.
364	514R	10/4/96	DS

SEARCH NOTES

	Date	Exmr.
APS	9/5/96	DS
MAYA	9/6/96	DS
STIC/EIC Search	9/11/96	DS
Andrew Faile 2611 — 455/72, 109, 47, 455/72, 38.1	9/30/96	DS

(RIGHT OUTSIDE)

```
=> d his
     (FILE 'USPAT' ENTERED AT 16:36:13 ON 05 SEP 96)
L1        427975 S COMPRESS#####
L2          9399 S VOICE(5A)SIGNAL#
L3           382 S L1 (10A) L2
L4         39488 S BANDWIDTH
L5          6057 S SUB(3W)CHANNEL####
L6            22 S L3 AND L5
L7       1310443 S TIME
L8         25713 S L1 (10A) L7
L9             5 S L8 AND L6
L10         3882 S SUB(2W)CHANNEL##### OR SUB()CHANNEL##### OR SUBCHANNEL##
###
L11           30 S L10 (P) L2
L12         1314 S SINGLE SIDEBAND MODULATION OR SSB
L13           14 S L10 (P) L12
L14            1 S L2 AND L13
L15            0 S L11 AND L12
L16       120674 S MODULAT#####
L17         1706 S L10 (P) (SIGNAL# OR L2)
L18           38 S L12 AND L17
L19           37 S L16 AND L18
L20         1869 S QUADRATURE (5A) AMPLITUDE#
L21            8 S L19 AND L20
=> s Waveform (5a) Similarity (10a) Overlap(5a)Add
         71661 WAVEFORM
         13458 SIMILARITY
         92176 OVERLAP
        171987 ADD
L22            0 WAVEFORM (5A) SIMILARITY (10A) OVERLAP(5A)ADD
=> s WSOLA
L23            0 WSOLA
```

File 347:JAPIO OCT 1976-1996/Apr.
(c) JPO & JAPIO
File 348:EUROPEAN PATENTS 1978-1996/SEP W1
(c) 1996 European Patent Office
File 350:Derwent World Pat. 1963-1980/UD=9632
(c) 1996 Derwent Info Ltd
File 351:DERWENT WPI 1981-1996/UD=9635;UA=9631;UM=9623
(c)1996 Derwent Info Ltd
?ds

Set	Items	Description
S1	227391	VOICE OR AUDIO OR ANALOG?
S2	72779	S1(2N)SIGNAL?
S3	1893201	COMPRESS? OR REDUC? OR REDN?
S4	2608	SUBCHANNEL? OR SUBCIRCUIT? OR SUBFACILIT? OR SUBLINE? OR S-UBPATH? OR SUBLINK? OR SUB()(CHANNEL? OR CIRCUIT? OR FACILIT? OR LINE? OR PATH? OR LINK?)
S5	1563136	TIME
S6	696	SINGLE(1W)(SIDEBAND? OR SIDE()BAND? ?)
S9	4	S1(4N)S3 AND S4 AND S5
S11	1	S1(4N)S3 AND S4 AND S6
S12	0	S9 AND S10

?t9/9/all

9/9/1 (Item 1 from file: 347)
DIALOG(R)File 347:JAPIO

04336768
CHANNEL MANAGEMENT SYSTEM AND MULTIPLEXING DEVICE IN ISDN SWITCHING NETWORK

PUB. NO.: 05-328468 [JP 5328468 A]
PUBLISHED: December 10, 1993 (19931210)
INVENTOR(s): MITADERA HIRONO
IWAMOTO SHIGENAGA
APPLICANT(s): NEC CORP [000423] (A Japanese Company or Corporation), JP (Japan)
NEC ENG LTD [329822] (A Japanese Company or Corporation), JP (Japan)
APPL. NO.: 04-148028 [JP 92148028]
FILED: May 15, 1992 (19920515)
INTL CLASS: [5] H04Q-011/04; H04J-003/16; H04L-012/52; H04M-003/00
JAPIO CLASS: 44.4 (COMMUNICATION -- Telephone); 44.2 (COMMUNICATION -- Transmission Systems); 44.3 (COMMUNICATION -- Telegraphy)
JOURNAL: Section: E, Section No. 1524, Vol. 18, No. 155, Pg. 52, March 15, 1994 (19940315)

ABSTRACT

PURPOSE: To improve the use efficiency of an information channel by multiplexing a call of plural voice system by using the same information channel, in the ISDN switching system.

CONSTITUTION: A multiplexing device 100 which is connected to an ISDN switching network 200 and contains analog voice terminals 301-308 is provided with a signal converting circuit 101 for converting an analog signal to a digital sound signal and an I. 451 call control signal or converting the digital sound signal and the I. 451 call signal to the analog signal, a band compression circuit 102 for allowing the

digital sound signal to be subjected to band compression to 16Kbps, a multiplexing/decomposing circuit 103 for multiplexing the signal subjected to band compression to a time slot in an information channel or decomposing it from in the time slot, a channel control part 104 for managing whether a sub - channel corresponding to the time slot is free or busy, and a concentrating/distributing circuit 105.

9/9/2 (Item 2 from file: 347)
DIALOG(R)File 347:JAPIO

02165917
LOGIC CIRCUIT

PUB. NO.: 62-082817 [JP 62082817 A]
PUBLISHED: April 16, 1987 (19870416)
INVENTOR(s): TAKAHASHI HIDEO
APPLICANT(s): NEC CORP [000423] (A Japanese Company or Corporation), JP (Japan)
APPL. NO.: 60-225261 [JP 85225261]
FILED: October 08, 1985 (19851008)
INTL CLASS: [4] H03K-019/00; H03K-019/094
JAPIO CLASS: 42.4 (ELECTRONICS -- Basic Circuits)
JAPIO KEYWORD:R097 (ELECTRONIC MATERIALS -- Metal Oxide Semiconductors, MOS)
JOURNAL: Section: E, Section No. 540, Vol. 11, No. 284, Pg. 145, September 12, 1987 (19870912)

ABSTRACT

PURPOSE: To reduce charging/discharging current and to reduce noise disturbance to an analog circuit by selecting a serial resistance value so that the states of (n) logical elements circuits to be subcircuits connected to a main circuit in parallel are successively changed with delays.

CONSTITUTION: When the change of an input signal from an inverter 2 is received by an inverter 6, a MOS transistor (TR) 9 is connected and electrostatic charge accumulated in a capacitor 7 starts to be discharged. On the other hand, the output signal from the inverter 2 is supplied to an inverter 4 through a serial resistor 3. The output of the inverter 4 is changed by a time delay determined by the resistance value of the serial resistor 3 and the input capacity of the inverter 4 to discharge the charge accumulated in the capacitor 5. When respective load current values of TRs 8, 9 are selected as a half of the capacity of an output buffer TR, the capacity of an input data becomes also a half, the ON resistance value of the inverters 4, 6 become about twice and the discharge peak current values of the capacitors 5, 7 become about a half. If the discharge is slightly shifted by inserting the serial resistor 3, noise due to discharge can be reduced to about a half.

9/9/3 (Item 3 from file: 347)
DIALOG(R)File 347:JAPIO

01147175
MULTIPLEX MODE SWITCHING DEVICE

PUB. NO.: 58-084575 [JP 58084575 A]

PUBLISHED: May 20, 1983 (19830520)
INVENTOR(s): AKATSUKA TERUMOTO
KAWAMOTO NOBORU
NAKAMURA JOJI
APPLICANT(s): MATSUSHITA ELECTRIC IND CO LTD [000582] (A Japanese Company or Corporation), JP (Japan)
APPL. NO.: 56-182821 [JP 81182821]
FILED: November 13, 1981 (19811113)
INTL CLASS: [3] H04N-005/40; H04N-005/60
JAPIO CLASS: 44.6 (COMMUNICATION -- Television)
JAPIO KEYWORD:R101 (APPLIED ELECTRONICS -- Video Tape Recorders, VTR); R102 (APPLIED ELECTRONICS -- Video Disk Recorders, VDR)
JOURNAL: Section: E, Section No. 191, Vol. 07, No. 178, Pg. 141, August 06, 1983 (19830806)

ABSTRACT

PURPOSE: To reduce noise caused in an audio multiplex modulator of a television receiver, in switching an audio multiplex modulator from the multiplex mode to the monaural mode, by providing a signal switching device controlling the passing of a control channel signal with a multiplex mode switching control signal.

CONSTITUTION: A signal switching device 20 has an input terminal 21 to which a control signal from an input terminal 9 is inputted and its control signal control the passing of a control channel signal 1. Noise caused when the mode is switched from the multiplex mode to the monaural mode, can be prevented by setting the delay time of a delay circuit 22 to a prescribed value or over. Inversely, in switching the mode from the monaural to the multiplex mode, since the output of multiplex signal of a multiplex modulator is definitely determined with the switching of the device 20, a sub - channel signal 2 and a control channel signal 1 are outputted from the multiplex modulator at the same time , it is quite out of question for the discrimination of the multiplex mode in, e.g., a television receiver.

9/9/4 (Item 1 from file: 351)
DIALOG(R)File 351:DERWENT WPI

009620124 WPI Acc No: 93-313673/40
XRPX Acc No: N93-241552
Time -division multiple access digital radio communication - splitting physical transmission channel into several logic channels dedicated to different communications
Index Terms: TIME DIVIDE MULTIPLE ACCESS DIGITAL RADIO COMMUNICATE SPLIT PHYSICAL TRANSMISSION CHANNEL LOGIC CHANNEL DEDICATE COMMUNICATE
Patent Assignee: (MATR-) MATRA COMMUNICATION
Author (Inventor): DELPRAT M; DORNSTETTER J; MEGE P
Number of Patents: 005
Number of Countries: 020
Patent Family:

Patent No	Kind	Date	Week	Applic No	Date	LA	Pages	IPC	
EP 564339	A1	931006	9340	EP 93400793	930326	Eng	16	H04J-003/16	(B)
FR 2689346	A1	931001	9348	FR 923883	920331			H04J-003/00	
CA 2092893	A	931001	9351	CA 2092893	930329			H04J-003/04	
FI 9301441	A	931001	9351	FI 931441	930330			H04J-000/00	
US 5398247	A	950314	9516	US 39729	930330		14	H04B-007/212	

Priority Data (CC No Date): FR 923883 (920331)

Applications (CC,No,Date): US 39729 (930330); EP 93400793 (930326); CA
2092893 (930329); FI 931441 (930330)
Language: English
EP and/or WO Cited Patents: 02Jnl.Ref; DE 2262933; EP 210698; GB 834247
Designated States
(Regional): AT; BE; CH; DE; DK; ES; GB; GR; IE; IT; LI; LU; MC; NL; PT; SE
Abstract (Basic): EP 564339 A

The communication method involves distributing a physical transmission channel into several logic channels. These consist of time slots with the same duration and rank in successive frames of constant length. One logic channel is divided into several sub - channels .

Specific groups of k successive time slots are assigned to each sub - channel . The groups are regularly distributed. N is not equal to (mk+1) where m is an integer and N is the number of time slots necessaryb to construct an information block.

ADVANTAGE - Decreases transmission delay. Reduced detrimental effect of localised perturbations.

Dwg.6/18

Abstract (US): 9516 US 5398247 A

The method involves successively transmitting TDM frames of constant length, each comprising a succession of p time slots of constant duration, so as to distribute a physical transmission channel into p logic channels. Each of the latter comprise time slots having a same rank in the TDMA frames. At least one of the logic channels is divided into n sub - channels dedicated to different communications, by assigning respective groups of k consecutive time slots pertaining to the logic channel to a respective one of the n sub - channels .

The groups of k consecutive time slots assigned to each of the n sub - channels are regularly time -distributed, formatting the information to be transmitted on a sub - channel as a sequence of information blocks each extending over N bursts.

USE/ADVANTAGE - In data or voice communication. Reduced transmission delays introduced by communication transmitted on logic sub - channels having reduced rate without increasing transmission delay for other sub - channels .

Dwg.1/18

File Segment: EPI
Derwent Class: W01; W02;
Int Pat Class: H04B-007/212; H04B-007/26; H04J-003/04; H04J-003/06;
H04J-003/16; H04L-005/22; H04Q-007/04
Manual Codes (EPI/S-X): W01-B05A1A; W02-C03C1A; W02-C03C3E; W02-K02
?t11/9/1

11/9/1 (Item 1 from file: 351)
DIALOG(R)File 351:DERWENT WPI

004283332 WPI Acc No: 85-110210/18
XRPX Acc No: N85-082708
Electronic security and surveillance system has sub - channel modulator using key frequency unique to each unit for audio and compressed video information and alarm code
Index Terms: ELECTRONIC SECURE SURVEILLANCE SYSTEM; SUB CHANNEL MODULATE KEY FREQUENCY UNIQUE UNIT AUDIO COMPRESS VIDEO INFORMATION ALARM CODE
Patent Assignee: (MICR-) MICRON INT LTD
Author (Inventor): RODRIQUEZ M J
Number of Patents: 001

```
Patent Family:
    CC Number    Kind   Date     Week
    US 4511886    A    850416    8518   (Basic)
Priority Data (CC No Date): US 538848 (831006); US 499946 (830601)
Abstract (Basic): US 4511886
```

Each remote installation includes surveillance equipment, including video, audio, and alarm signals, associated with monitored locations. The security information collected by the surveillance equipment is serially sampled by a switcher which provides that information to an interface unit transmitter. The interface unit transmitter compresses the video information and decodes the alarm information and using a key frequency and single side band modulation techniques modulates and sub - channelises the processed security information into a frequency spectrum.

The sub - channelised security information is translated in frequency and transmitted on the transmission medium. The information received at the central station is demodulated, and the alarm information monitored by a command computer. The system provides an upstream command channel so that the central station can communicate with each remote installation.

ADVANTAGE - Transmission bandwidth is conserved. @(17pp Dwg.No.1/7

```
File Segment: EPI
Derwent Class: W02; W05;, R31; R57;
Int Pat Class: G08B-001/08; H04N-007/18
Manual Codes (EPI/S-X): W02-F01; W05-B05
?
```

```
Set     Items   Description
S1     227391   VOICE OR AUDIO OR ANALOG?
S2      72779   S1(2N)SIGNAL?
S3    1893201   COMPRESS? OR REDUC? OR REDN?
S4       2608   SUBCHANNEL? OR SUBCIRCUIT? OR SUBFACILIT? OR SUBLINE? OR S-
             UBPATH? OR SUBLINK? OR SUB()(CHANNEL? OR CIRCUIT? OR FACILIT?
             OR LINE? OR PATH? OR LINK?)

S13         1   (BANDWID? OR BAND(1N)WIDTH?)(4N)S3 AND S4
S14         1   (VOCOD? OR COMPANDOR?) AND S4

S16         2   S13 OR S14
?
t16/9/all
```

16/9/1   (Item 1 from file: 347)
DIALOG(R)File 347:JAPIO

02707638
2ND SOUND MULTIPLEX BROADCAST EQUIPMENT FOR FM BROADCAST

PUB. NO.: 01-005238 [JP 1005238 A]
PUBLISHED: January 10, 1989 (19890110)
INVENTOR(s): ITO AKIRA
MATSUURA TAKASHI
APPLICANT(s): NEC CORP [000423] (A Japanese Company or Corporation), JP (Japan)
APPL. NO.: 62-162108 [JP 87162108]
FILED: June 29, 1987 (19870629)
INTL CLASS: [4] H04J-001/00
JAPIO CLASS: 44.2 (COMMUNICATION -- Transmission Systems)
JOURNAL: Section: E, Section No. 749, Vol. 13, No. 176, Pg. 55, April 25, 1989 (19890425)

ABSTRACT

PURPOSE: To send voice and information with a large capacity by using a binary or a quaternary PSK so as to modulate a 2nd subcarrier whose upper limit is set to nearly middle of the occupied band width of FM wave as the coded voice signal and superimposing the result.

CONSTITUTION: A PSK modulator 3 multiplies a 19kHz of a pilot signal from a stereo modulator 4 by four to generate the 2nd subcarrier of a frequency of 76kHz and the result is subject to PSK modulation by a 2nd voice signal coded by a VOCODER 2. The output of the stereo modulator 4 and the output of the PSK modulator 3 are synthesized by a signal synthesis section 7. The frequency of the 2nd subcarrier is formed such that its lower limit is selected by the upper limit frequency of the stereo subchannel and its upper limit is limited by the specification of the occupied band width and its frequency is nearly a middle frequency of the frequency band and the PSK modulation of the 2nd subcarrier by the 2nd voice is applied by utilizing the frequency band sufficiently. Thus, the transmission information quantity is increased than a conventional system.

16/9/2   (Item 1 from file: 351)
DIALOG(R)File 351:DERWENT WPI

008525438 WPI Acc No: 91-029522/04

RPX Acc No: N91-022730 *Image available*
High definition television receiver with picture-in-picture display - receives signals subjected to bandwidth compression by sub-nyquist sampling which are reduced by time base compression
ndex Terms: HIGH DEFINE TELEVISION PICTURE PICTURE DISPLAY RECEIVE; RECEIVE SIGNAL SUBJECT BANDWIDTH COMPRESS SUB NYQUIST SAMPLE REDUCE TIME BASE COMPRESS
'atent Assignee: (MATU) MATSUSHITA ELEC IND KK
uthor (Inventor): ISOBE M; HAMADA
Iumber of Patents: 001
'atent Family:

CC Number	Kind	Date	Week	
US 4982288	A	910101	9104	(Basic)

'riority Data (CC No Date): US 310925 (890216)
'iling Details: US4982288 (1810RMC)
ıbstract (Basic): US 4982288

The appts. is of the type in which when performing a picture in picture operation by receiving picture signals compressed in bandwidth by multiple sub-Nyquist sampling, a sub - channel signal is subjected to a spatial interpolating process and combined with a main-channel signal. The still picture portion and moving picture portion of one input signal are restored to a field offset sub-sampled first picture signal.

A second input signal is restored to a field offset sub-sampled picture signal, subjected to a size-reducing process by time base compression in the vertical and horizontal directions of the picture and delivered as a second picture signal. This signal of the form synchronised in phase with a given position of the first picture signal. Both picture signals are time-division multiplexed to deliver a third picture signal onto a picture screen.

ADVANTAGE - Overall simplified construction of receiver. @(7pp Dwg.No.1/3

File Segment: EPI
Derwent Class: W03; R57;
Int Pat Class: H04N-005/27
Manual Codes (EPI/S-X): W03-A11; W03-A13
?

Set	Items	Description
S1	227391	VOICE OR AUDIO OR ANALOG?
S2	72779	S1(2N)SIGNAL?
S3	1893201	COMPRESS? OR REDUC? OR REDN?
S5	1563136	TIME
S6	696	SINGLE(1W)(SIDEBAND? OR SIDE()BAND? ?)
S17	10574	BASESTATION? OR LANDSTATION? OR (BASE OR LAND)()STATION?
S18	19	S17 AND S1(4N)S3 AND S5
S19	1	S17 AND S1(4N)S3 AND S6
S20	1	S18 AND S19

?
t20/9/1

20/9/1 (Item 1 from file: 351)
DIALOG(R)File 351:DERWENT WPI

003843945 WPI Acc No: 83-840195/50
Related WPI Accession(s): 88-021377
XRPX Acc No: N83-221411
Two-way SSB land mobile RF communication system has pilot tone signal transmitted with audio signal to receiver, having PLL to acquire; PHASE LOCK LOOP SINGLE SIDE BAND RADIO FREQUENCY
Index Terms: TWO WAY SSB LAND MOBILE RF COMMUNICATE SYSTEM PILOT TONE SIGNAL TRANSMIT AUDIO SIGNAL RECEIVE PLL ACQUIRE
Patent Assignee: (SIDE-) SIDEBAND TECH INC; (AERO-) AEROTRON INC; (GESJ) GENERAL SIGNAL CORP
Author (Inventor): JACOBS P H; COLLETTE D P
Number of Patents: 013
Patent Family:

CC Number	Kind	Date	Week	
EP 95685	A	831207	8350	(Basic)
GB 2121251	A	831214	8350	
JP 59036433	A	840228	8414	
US 4539707	A	850903	8538	
CA 1194130	A	850924	8543	
US 4573208	A	860225	8611	
GB 2167273	A	860521	8621	
GB 2167273	B	870107	8701	
GB 2121251	B	870114	8702	
IL 68809	A	871231	8809	
IL 83964	A	880630	8835	
EP 95685	B	890809	8932	
DE 3380389	G	890914	8938	

Priority Data (CC No Date): US 384148 (820601); US 558046 (840126); US 807641 (851211)
Applications (CC,No,Date): JP 8395883 (830601); GB 8531266 (851220); GB 8314950 (830531); EP 83105045 (830520)
Language: English
EP and/or WO Cited Patents: No.SR.Pub; A3 ...8627; FR 2159187; US 3217255; FR 1226996; EP 44374; DE 2937986; US 3108158; JP 53054404; US 4300237; 1.Jnl.REF
Designated States
(Regional): CH; DE; IT; LI; NL
Filing Details: US4573208 Div.ex 4539707 (904MN); GB2167273 Derived from 31.05.83 014950
Abstract (Basic): The system has a push-to-talk transmitter which broadcasts a compressed signal for a receiver. The compressed signal

includes an unmodulated frequency modulated pilot tone during an initial predetermined time interval. After this interval, the compressed signal includes a composite signal having an attenuated frequency modulated pilot tone and an audio signal. The transmitter gain is adjusted to produce full rated power during the predetermined interval and is unadjusted after that.

The receiver has a detector for the compressed composite signal, from which the pilot tone signal is separated by a filter. A phase lock loop responds to a locally generated frequency modulated tone and the pilot tone filter to vary the filter characteristics of the PLL in order to adjust the frequency of the detected compressed composite signal. The PLL also varies the filter characteristics of the pilot tone filter to adjust the receiver gain. The audio signal is filtered from the compressed composite signal which is expanded and amplified, in response to the pilot tone filter, without doing the same to the pilot tone. (51pp Dwg.No.0/10

Abstract (US): 8611 US 4573208

A pilot tone is transmi-ted at full rated power to aid in acquisition of a signal by the receiver. The transmitter ALC is then disabled and the pilot tone attenuated. The receiver adjusts the frequency characteristics of the pilot tone and phase lock loop filters in the detector as a function of lock on.

The delay after loss of lock-on in reverting to wide band pilot tone and wideband loop filters is varied as a function of signal strength. The pilot tone may be modulated for tone coded squelch. The modulating source is located in the return end of the PLL filter. Automatic gain control of audio signal is responsive to tone signal without affecting composite audio and tone signal. ADVANTAGE - The latter is formed from an underdamped, band-pass filter with a peak at the high end of the pass-band.@(13pp)@ 8538 US 4539707

The audio signal is compressed prior to pre-emphases and then summed with a pilot tone for further compression prior to transmission. Initially, only the pilot tone is transmitted at full rated power to aid in acquisition of the signal by the receiver. The transmitter ALC is then disabled and the pilot tone is attenuated. The receiver adjusts the frequency characteristics of the pilot tone filter and phase lock loop filter in the detector as a function of lock-on.

The delay after loss of lock-on in reverting to wide band pilot tone and wideband loop filters is varied as a function of signal strength. The pilot tone may be modulated for tone coded squelch. The modulating source is located in the return end of the phase lock loop filter. A unique filter is provided to insure acquisition of the pilot tone. Automatic gain control of the audio signal is responsive to the tone signal without effecting the compositite audio and tone signal. @(14pp

Abstract (GB): 8702 GB 2121251

A method of enhancing reception of a composite radio frequency signal including audio frequency components and a pilot tone component by a receiver having a radio frequency oscillator, a phase lock loop, and radio frequency and intermediate frequency amplifiers in which (a) the pilot tone component is detected by a pilot tone filter and rectifier means and the amplitude of the detected pilot tone component is used to control the gain of the radio frequency and intermediate frequency amplifiers within the receiver, (b) and audio frequency components are de-emphasised for application to a speaker, and (c) the phase lock loop receives both the detected pilot tone component and a locally generated pilot tone and adjusts the radio frequency oscillator for frequency errors in the received carrier frequency, wherein the frequency response characteristics of the pilot tone filter are selectively varied as a function of phase lock loop lock-on. 8701 GB 2167273

A method of transmitting from a radiotransmitter a modulation signal, wherein the modulation signal is a composite audio signal including audio signal components and a pilot tone component above the aduio signal passband but within the bandpass of the transmitter, and wherein the modulation signal contains solely the pilot tone component for a predetermined initial _time_ interval following the initiation of a transmission to aid in the acquisition of the transmitted signal by a receiver.

Abstract (EP): 8932 EP 95685

RF communications method in which a particular combination of a pilot and an audio signal is transmitted at the transmitter and in the receiver the particular combination of audio and pilot is detected, whereby the pilot is used to synchronize a local oscillator of the receiver and also to control reception of the accompanying _audio_ comprising broadcasting a _compressed_ signal from a _base_ _station_ by a push-to-talk _single_ _sideband_ , radio communication transmitter, said compressed signal comprising only a frequency modulated pilot tone during an initial predetermined _time_ interval and thereafter a composite signal including a frequency modulated pilot tone and an audio signal, characterised in that the output power of said transmitter is adjusted to full rated transmitter power during said initial predetermined _time_ interval with the pilot tone being unattenuated and in that the gain of said transmitter is maintained constant thereafter with the pilot tone being attenuated. @(19pp)@

File Segment: EPI

Derwent Class: W02; R56; R54; R58; R55

Int Pat Class: H03G-003/20; H04B-001/68; H04B-007/26; H04Q-007/02; H03L-007/06

Manual Codes (EPI/S-X): W02-C03X; W02-G02; W02-G04

?

```
Set     Items   Description
S1      227391  VOICE OR AUDIO OR ANALOG?
S2       72779  S1(2N)SIGNAL?
S3     1893201  COMPRESS? OR REDUC? OR REDN?

S5     1563136  TIME
S6         696  SINGLE(1W)(SIDEBAND? OR SIDE()BAND? ?)

S21       1085  S1(4N)S3 AND RECEIV?
S22        356  S21 AND S5
S23          8  S21 AND S6
S24          3  S22 AND S23
?
t24/9/all
```

24/9/1 (Item 1 from file: 350)
DIALOG(R)File 350:Derwent World Pat.

002391180 WPI Acc No: 80-K7650C/45
Transceiver AM radio telephony - has mean amplitude of voice input coded digitally into transmitted signals
Index Terms: TRANSCEIVER AM RADIO TELEPHONE MEAN AMPLITUDE VOICE INPUT CODE DIGITAL TRANSMIT SIGNAL
Patent Assignee: (CSFC) THOMSON-CSF
Author (Inventor): DEMAN P; PIMENTEL A; BENSADOU J C; DERIVEIERE C
Number of Patents: 008
Patent Family:

CC Number	Kind	Date	Week	
EP 18256	A	801029	8045	(Basic)
BR 8002229	A	801202	8051	
FR 2454231	A	801212	8106	
US 4326293	A	820420	8218	
IL 59800	A	820730	8234	
CA 1149463	A	830705	8330	
EP 18256	B	841010	8441	
DE 3069399	G	841115	8447	

Priority Data (CC No Date): FR 799491 (790413)
Language: French
EP and/or WO Cited Patents: US 3882458; CH 461585; DE 2128889; FR 2258840; FR 2260334; 1.Jnl.REF
Designated States
(Regional): DE; GB; IT; NL; SE
Abstract (Basic): The transceiver for A.M. Single sideband suppressed carrier radio transmission has automatic voice switching between Transmit and Receive modes. Each time a period of speech activity is detected in the transmitter, a short delay is imposed on the speech signal while its mean amplitude is measured. This measurement is carried out digitally, the analogue speech signal being converted in an A/D converter, stored in a memory and analysed in a digital processor.

The mean speech amplitude is coded digitally, and this code is added as a preamble to the analogue signal. The combined digital and analogue signals are amplitude compressed before the carrier is modulated and R.F. amplified. In the receiver , the digital code is decoded and used to control the setting of an attenuator, so restoring the amlitude of the analogue signal in the receiver to a level corresponding to that of the input to the transmitter.
File Segment: EPI
Derwent Class: W02; R56;

Int Pat Class: H04B-001/46
Manual Codes (EPI/S-X): W02-G02

24/9/2 (Item 1 from file: 351)
DIALOG(R)File 351:DERWENT WPI

003843945 WPI Acc No: 83-840195/50
Related WPI Accession(s): 88-021377
XRPX Acc No: N83-221411
Two-way SSB land mobile RF communication system has pilot tone signal transmitted with audio signal to receiver, having PLL to acquire; PHASE LOCK LOOP SINGLE SIDE BAND RADIO FREQUENCY
Index Terms: TWO WAY SSB LAND MOBILE RF COMMUNICATE SYSTEM PILOT TONE SIGNAL TRANSMIT AUDIO SIGNAL RECEIVE PLL ACQUIRE
Patent Assignee: (SIDE-) SIDEBAND TECH INC; (AERO-) AEROTRON INC; (GESJ) GENERAL SIGNAL CORP
Author (Inventor): JACOBS P H; COLLETTE D P
Number of Patents: 013
Patent Family:

CC Number	Kind	Date	Week	
EP 95685	A	831207	8350	(Basic)
GB 2121251	A	831214	8350	
JP 59036433	A	840228	8414	
US 4539707	A	850903	8538	
CA 1194130	A	850924	8543	
US 4573208	A	860225	8611	
GB 2167273	A	860521	8621	
GB 2167273	B	870107	8701	
GB 2121251	B	870114	8702	
IL 68809	A	871231	8809	
IL 83964	A	880630	8835	
EP 95685	B	890809	8932	
DE 3380389	G	890914	8938	

Priority Data (CC No Date): US 384148 (820601); US 558046 (840126); US 807641 (851211)
Applications (CC,No,Date): JP 8395883 (830601); GB 8531266 (851220); GB 8314950 (830531); EP 83105045 (830520)
Language: English
EP and/or WO Cited Patents: No.SR.Pub; A3 ...8627; FR 2159187; US 3217255; FR 1226996; EP 44374; DE 2937986; US 3108158; JP 53054404; US 4300237; 1.Jnl.REF
Designated States
(Regional): CH; DE; IT; LI; NL
Filing Details: US4573208 Div.ex 4539707 (904MN); GB2167273 Derived from 31.05.83 014950
Abstract (Basic): The system has a push-to-talk transmitter which broadcasts a compressed signal for a receiver. The compressed signal includes an unmodulated frequency modulated pilot tone during an initial predetermined time interval. After this interval, the compressed signal includes a composite signal having an attenuated frequency modulated pilot tone and an audio signal. The transmitter gain is adjusted to produce full rated power during the predetermined interval and is unadjusted after that.

The receiver has a detector for the compressed composite signal, from which the pilot tone signal is separated by a filter. A phase lock loop responds to a locally generated frequency modulated tone and the pilot tone filter to vary the filter characteristics of the PLL in order to adjust the frequency of the detected compressed

composite signal. The PLL also varies the filter characteristics of the pilot tone filter to adjust the receiver gain. The audio signal is filtered from the compressed composite signal which is expanded and amplified, in response to the pilot tone filter, without doing the same to the pilot tone. (51pp Dwg.No.0/10)

Abstract (US): 8611 US 4573208

A pilot tone is transmi-ted at full rated power to aid in acquisition of a signal by the receiver . The transmitter ALC is then disabled and the pilot tone attenuated. The receiver adjusts the frequency characteristics of the pilot tone and phase lock loop filters in the detector as a function of lock on.

The delay after loss of lock-on in reverting to wide band pilot tone and wideband loop filters is varied as a function of signal strength. The pilot tone may be modulated for tone coded squelch. The modulating source is located in the return end of the PLL filter. Automatic gain control of audio signal is responsive to tone signal without affecting composite audio and tone signal. ADVANTAGE - The latter is formed from an underdamped, band-pass filter with a peak at the high end of the pass-band.@(13pp)@ 8538 US 4539707

The audio signal is compressed prior to pre-emphases and then summed with a pilot tone for further compression prior to transmission. Initially, only the pilot tone is transmitted at full rated power to aid in acquisition of the signal by the receiver . The transmitter ALC is then disabled and the pilot tone is attenuated. The receiver adjusts the frequency characteristics of the pilot tone filter and phase lock loop filter in the detector as a function of lock-on.

The delay after loss of lock-on in reverting to wide band pilot tone and wideband loop filters is varied as a function of signal strength. The pilot tone may be modulated for tone coded squelch. The modulating source is located in the return end of the phase lock loop filter. A unique filter is provided to insure acquisition of the pilot tone. Automatic gain control of the audio signal is responsive to the tone signal without effecting the compositite audio and tone signal. @(14pp

Abstract (GB): 8702 GB 2121251

A method of enhancing reception of a composite radio frequency signal including audio frequency components and a pilot tone component by a receiver having a radio frequency oscillator, a phase lock loop, and radio frequency and intermediate frequency amplifiers in which (a) the pilot tone component is detected by a pilot tone filter and rectifier means and the amplitude of the detected pilot tone component is used to control the gain of the radio frequency and intermediate frequency amplifiers within the receiver , (b) and audio frequency components are de-emphasised for application to a speaker, and (c) the phase lock loop receives both the detected pilot tone component and a locally generated pilot tone and adjusts the radio frequency oscillator for frequency errors in the received carrier frequency, wherein the frequency response characteristics of the pilot tone filter are selectively varied as a function of phase lock loop lock-on. 8701 GB 2167273

A method of transmitting from a radiotransmitter a modulation signal, wherein the modulation signal is a composite audio signal including audio signal components and a pilot tone component above the aduio signal passband but within the bandpass of the transmitter, and wherein the modulation signal contains solely the pilot tone component for a predetermined initial time interval following the initiation of a transmission to aid in the acquisition of the transmitted signal by a receiver .

Abstract (EP): 8932 EP 95685

RF communications method in which a particular combination of a pilot and an audio signal is transmitted at the transmitter and in the receiver the particular combination of audio and pilot is detected, whereby the pilot is used to synchronize a local oscillator of the receiver and also to control reception of the accompanying audio comprising broadcasting a compressed signal from a base station by a push-to-talk single sideband, radio communication transmitter, said compressed signal comprising only a frequency modulated pilot tone during an initial predetermined time interval and thereafter a composite signal including a frequency modulated pilot tone and an audio signal, characterised in that the output power of said transmitter is adjusted to full rated transmitter power during said initial predetermined time interval with the pilot tone being unattenuated and in that the gain of said transmitter is maintained constant thereafter with the pilot tone being attenuated. @(19pp)@

File Segment: EPI
Derwent Class: W02; R56; R54; R58; R55
Int Pat Class: H03G-003/00; H04B-001/68; H04B-007/26; H04Q-007/02; H03L-007/06
Manual Codes (EPI/S-X): W02-C03X; W02-G02; W02-G04

24/9/3 (Item 2 from file: 351)
DIALOG(R)File 351:DERWENT WPI

003811851 WPI Acc No: 83-808095/44
XRPX Acc No: N83-197590
Single sideband analog transmission system has input audio signals time - compressed before modulation and and received signals time -expanded after demodulation; SSB
Index Terms: SINGLE SIDEBAND ANALOGUE TRANSMISSION SYSTEM INPUT AUDIO SIGNAL TIME COMPRESS MODULATE SIGNAL TIME EXPAND AFTER DEMODULATE RECEIVE
Patent Assignee: (GRIN-) GRINAKER ELTRN HOLD
Author (Inventor): COCCIUTI A S G; VANLOCHEM G W
Number of Patents: 002
Patent Family:

CC Number	Kind	Date	Week	
ZA 8206846	A	830620	8344	(Basic)
ZA 8206846	A	830620	8337	

Priority Data (CC No Date): ZA 816500 (810918); ZA 826846 (820917)
Abstract (Basic): The system includes a signal processor having two parallel charge-coupled devices (50,52) connected in parallel to an output OR gate (54). The input signals are clocked in at relatively low frequency and clocked out at relatively high frequency. When one of the devices is full the input signals are fed to the other. The output rate is higher to provide time compressed signals for modulation and transmission. For processing received signals the charge-coupled devices (56,58) are arranged for clocking in signals at relatively high frequency and for clocking them out at audio frequency.

In another form of the processor, the analog input signals for transmission and analog signals received are digitised prior to processing and are then converted back to analog signals again. The system is less susceptible to frequency drift and interference between the various channels is eliminated because each transmission occurs discretely in time for each channel. (Provisional Basic advised week 83/37) (10pp Dwg.No.2/3)
File Segment: EPI
Derwent Class: W02; W04; R56;

Int Pat Class: H04B-000/00
Manual Codes (EPI/S-X): W02-G02; W02-G04; W04-G
?

```
Set    Items   Description
S1    227391   VOICE OR AUDIO OR ANALOG?
S2     72779   S1(2N)SIGNAL?
S3   1893201   COMPRESS? OR REDUC? OR REDN?

S6       696   SINGLE(1W)(SIDEBAND? OR SIDE()BAND? ?)

S21     1085   S1(4N)S3 AND RECEIV?

S23        8   S21 AND S6

S25   112785   DEMODULAT? OR DE()MODULAT? OR MODEM? OR S1(3N)DIGIT?
S26        6   S23 AND S25
S27        4   S26 NOT S24
?
t27/9/all
```

27/9/1 (Item 1 from file: 348)
DIALOG(R)File 348:EUROPEAN PATENTS

00394397

System and method of frequency calibration in a linked compression-expansion (Lincompex) system.

PATENT ASSIGNEE:
AMAF INDUSTRIES, INC., (1098021), 9052 Old Annapolis Road, Columbia Maryland 21045, (US), (applicant designated states: AT;BE;CH;DE;DK;FR;GB;IT;LI;LU;NL;SE)
AUTHOR (Inventor):
Leveque, James Howard, 10094 Colonial Drive, Ellicott City, Maryland 21043, (US)
LEGAL REPRESENTATIVE:
Vossius & Partner (100311), Siebertstrasse 4 P.O. Box 86 07 67, D-8000 Munchen 86, (DE)
PATENT (CC, No, Kind, Date): EP 401850 A2 901212 (Basic)
EP 401850 A3 920603
APPLICATION (CC, No, Date): EP 90110878 900608;
PRIORITY DATA (CC, No, Date): US 363863 890609
LANGUAGE (Publication,Procedural,Application): English; English; English
DESIGNATED STATES: AT; BE; CH; DE; DK; FR; GB; IT; LI; LU; NL; SE
INTL PAT CLASS: H04B-001/64;
CITED PATENTS (EP A): US 4271499 A; US 4457020 A; EP 274958 A
WORD COUNT: 156

ABSTRACT: EP 401850 A2
A linked compressor-expander (Lincompex) circuit for use in telecommunications utilizes a frequency compensation circuit (56) to frequency shift the total bandwidth of the whole communication channel to eliminate single side band "duck-talk" and data distortion in the transmitting of a complex waveform due to frequency drift or the detuning of the transmitter/ receiver system. The demodulator includes an expander (42) for expanding the compressed voice or data signal and a control tone conversion circuit (27) for converting the received control tone into a frequency signal to be used by the expander (42). The control tone conversion circuit (27) also determines the frequency error of a combination information waveform so that the frequency compensation circuit (56) can carry out the proper frequency shifting process. A fading regulator (58) is also provided to remove any audible level variations not removed by the automatic gain control circuitry in the receiver (38). (see image in original document)

```
LEGAL STATUS (Type, Pub Date, Kind, Text):
 Application:      901212 A2 Published application (A1withSR;A2withoutSR)
 Search Report:    920603 A3 Separate publication of the European or
                             International search report
 Examination:      930707 A2 Date of filing of request for examination:
                             930505
 Examination:      941214 A2 Date of despatch of first examination report:
                             941026
*Assignee:         950719 A2 Applicant (transfer of rights) (change): LINK
                             PLUS CORPORATION (1098022) 32 Loockerman
                             Square, Suite L-100 City of Dover, County of
                             Kent (US) (applicant designated states:
                             AT;BE;CH;DE;DK;FR;GB;IT;LI;LU;NL;SE)
*Assignee:         950719 A2 Previous applicant in case of transfer of
                             rights (change): AMAF INDUSTRIES, INC.
                             (1098021) 9052 Old Annapolis Road Columbia
                             Maryland 21045 (US) (applicant designated
                             states: AT;BE;CH;DE;DK;FR;GB;IT;LI;LU;NL;SE)
 Withdrawal:       960605 A2 Date on which the European patent application
                             was deemed to be withdrawn: 951130
```

```
 27/9/2     (Item 1 from file: 351)
DIALOG(R)File 351:DERWENT WPI
```


```
008482899  WPI Acc No: 90-369899/50
XRPX Acc No: N90-282055
    Frequency calibration in compression-expansion system - compressed
    signal  receiver  detects frequency errors, then compensates expanded
    output and controls stone frequencies; LINK
Index Terms: FREQUENCY CALIBRATE COMPRESS EXPAND SYSTEM COMPRESS SIGNAL
     RECEIVE  DETECT FREQUENCY ERROR COMPENSATE EXPAND OUTPUT CONTROL
    STONE FREQUENCY
Patent Assignee: (AMAF-) AMAF IND INC; (AMAF-) AMAF INDS INC; (LEVE/)
    LEVEQUE J H; (LINK-) LINK PLUS CORP
Author (Inventor): LEVEQUE J H
Number of Patents: 007
Number of Countries: 015
Patent Family:
Patent No Kind  Date  Week Applic No    Date  LA  Pages IPC
EP 401850   A  901212 9050 EP 90110878 900608                        (B)
AU 9056805  A  901213 9106
CA 2018174  A  901209 9110
JP 3029529  A  910207 9112 JP 90152528 900611
US 5065451  A  911112 9148 US 363863   890609
EP 401850   A3 920603 9332 EP 90110878 900608
AU 642880   B  931104 9351 AU 9056805  900605          H04B-001/64

Priority Data (CC No Date): US 363863 (890609)
Applications (CC,No,Date): EP 90110878 (900608); JP 90152528 (900611); EP
    90110878 (900608); AU 9056805 (900605)
Language: English
EP and/or WO Cited Patents: NoSR.Pub; EP 274958   A; US 4271499   D; US
    4457020   A
Designated States
 (Regional): AT; BE; CH; DE; FR; GB; IT; LI; LU; NL; SE
Filing Details: AU0642880 Previous Publ.   AU 9056805
Abstract (Basic): EP 401850    A
```

A receiver receives compressed combinational information which consists of the wanted signal and a control tone. These are separated by an input filter. The compressed signal is expanded to receover the original one. The frequency error between this and the transmitted signal is calculated.

A frequency compensation network uses the error to correct the expanded signal and control tone frequencies.

USE - Lincomplex telecommunication circuits. @(20pp Dwg.No.5/8

Abstract (US): 9148 US 5065451

The compressor-expander (Lincompex) circuit utilises a frequency compensation circuit to frequency shift the total bandwidth of the whole communication channel. The demodulator includes an expander for expanding the compressed voice or data signal and a control tone conversion circuit for converting the received control tone into a frequency signal to be used by the expander.

The control tone conversion circuit also determines the frequency error of a combination information waveform so that the frequency compensation circuit can carry out the proper frequency shifting process. A fading regulator is also provided to remove any audible level variations not removed by the automatic gain control circuitry in the receiver .

USE - Eliminates single side band "duct-talk" and data distortion in the transmitting of a complex waveform due to . frequency drift or detuning of transmitter/ receiver system. @(20pp)@

File Segment: EPI
Derwent Class: U24; W02;
Int Pat Class: H03G-007/00; H04B-001/64; H04B-007/00; H04B-014/00
Manual Codes (EPI/S-X): U24-C02B; W02-G03X; W02-G04

27/9/3 (Item 2 from file: 351)
DIALOG(R)File 351:DERWENT WPI

007638688 WPI Acc No: 88-272620/39
XRPX Acc No: N88-207078 *Image available*
Amplitude modulator using high frequency feedback - has detected high frequency modulator output fed back to low frequency input automatic output control circuit; SSB SINGLE SIDEBAND RADIO TRANSCEIVER
Index Terms: AMPLITUDE MODULATE HIGH FREQUENCY FEEDBACK; DETECT HIGH FREQUENCY MODULATE OUTPUT FEED BACK LOW FREQUENCY INPUT AUTOMATIC OUTPUT CONTROL CIRCUIT
Patent Assignee: (PHIG) PHILIPS GLOEILAMPEN NV; (NIMA-) NIHON MARANZ KK
Author (Inventor): KATSUHIRO E; YASUMICHI I
Number of Patents: 003
Patent Family:

CC Number	Kind	Date	Week	
EP 284152	A	880928	8839	(Basic)
JP 63232506	A	880928	8845	
US 4866405	A	890912	8946	

Priority Data (CC No Date): JP 8763698 (870320)
Applications (CC,No,Date): EP 88200506 (880318); US 169676 (880318)
Language: English
EP and/or WO Cited Patents: A3...9026; EP 95685; FR 2221859; US 4088956
Designated States
(Regional): DE; FR; GB
Abstract (Basic): EP 284152

The modulator compresses the dynamic range of an audio signal by a compression . type amplifier (1). The amplifier output is passed via an automatic output control circuit (2) to a balanced high frequency modulation circuit (3) for mixing with a carrier wave oscillator (4) output.

A portion of the high frequency output of the modulator, which varies linearly with the audio imput, is amplified (5) and detected (6) to provide a DC signal. This signal is fed to the automatic output control circuit (2) to control the amplitude of the low frequency signal.

USE/ADVANTAGE - SSB radio transceiver. Provides good and low distortion. @(5pp Dwg.No.1/1

Abstract (US): 8946 US 4866405

The modulation appts. has an automatic output control device for receiving an input signal corresponding to the low frequency signal and for producing a contorlled low frequency signal. A high frequency signal is also produced. The high frequency signal is modulated with the controlled low frequency signal to produce the modulated high frequency signal. A discriminator receives and demodulates at least a portion of the modulated high frequency signal and produces a feedback signal that corresponds only to the modulation of the modulated high frequency signal and, thus, also corresponds only to the low frequency signal. The automatic control device receives the feedback signal and controls the amplitude of the controlled low frequency signal in response to the feedback signal. @(4pp)@

File Segment: EPI

Derwent Class: U23; U24; W02; R54;

Int Pat Class: H03C-001/02

Manual Codes (EPI/S-X): U23-G; U24-C02B; W02-G02; W02-G04

27/9/4 (Item 3 from file: 351)

DIALOG(R)File 351:DERWENT WPI

004283332 WPI Acc No: 85-110210/18

XRPX Acc No: N85-082708

Electronic security and surveillance system has sub-channel modulator using key frequency unique to each unit for audio and compressed video information and alarm code

Index Terms: ELECTRONIC SECURE SURVEILLANCE SYSTEM; SUB CHANNEL MODULATE KEY FREQUENCY UNIQUE UNIT AUDIO COMPRESS VIDEO INFORMATION ALARM CODE

Patent Assignee: (MICR-) MICRON INT LTD

Author (Inventor): RODRIQUEZ M J

Number of Patents: 001

Patent Family:

CC Number	Kind	Date	Week	
US 4511886	A	850416	8518	(Basic)

Priority Data (CC No Date): US 538848 (831006); US 499946 (830601)

Abstract (Basic): US 4511886

Each remote installation includes surveillance equipment, including video, audio, and alarm signals, associated with monitored locations. The security information collected by the surveillance equipment is serially sampled by a switcher which provides that information to an interface unit transmitter. The interface unit transmitter compresses the video information and decodes the alarm information and using a key frequency and single side band modulation techniques modulates and sub-channelises the processed security information into a frequency spectrum.

The sub-channelised security information is translated in frequency and transmitted on the transmission medium. The information received at the central station is demodulated , and the alarm information monitored by a command computer. The system provides an upstream command channel so that the central station can communicate with each remote installation.

ADVANTAGE - Transmission bandwidth is conserved. @(17pp Dwg.No.1/7

File Segment: EPI
Derwent Class: W02; W05; R31; R57;
Int Pat Class: G08B-001/08; H04N-007/18
Manual Codes (EPI/S-X): W02-F01; W05-B05
?

```
Set     Items   Description
S1     227391   VOICE OR AUDIO OR ANALOG?
S2      72779   S1(2N)SIGNAL?

S4       2608   SUBCHANNEL? OR SUBCIRCUIT? OR SUBFACILIT? OR SUBLINE? OR S-
             UBPATH? OR SUBLINK? OR SUB()(CHANNEL? OR CIRCUIT? OR FACILIT?
             OR LINE? OR PATH? OR LINK?)
S5    1563136   TIME

S28     36319   PAGE OR RADIOPAG? OR PAGING OR BEEP? OR PAGE? ? OR RADIO? -
             ?(2N)(SUMMON? OR PRECONI?)
S29        20   S28 AND S4
S30         0   S29 AND AMPLI?
S31         0   S29 AND FILTER?
S32         8   S29 AND S5
S33         8   S32 NOT (S9 OR S11 OR S16 OR S20 OR S24 OR S27)
S34         0   S33 AND S1(4N)S3
S35         0   S33 AND S2
S36         2   S33 AND S1
?
t36/9/all

 36/9/1     (Item 1 from file: 351)
DIALOG(R)File 351:DERWENT WPI
(c)1996 Derwent Info Ltd. All rts. reserv.

009858045  WPI Acc No: 94-137901/17
XRPX Acc No: N94-108323      *Image available*
    Frequency changing method between adjacent cells in TDMA
    telecommunications - using many  voice  channels changing to new
    carrier in sequence with signalling channel transfer within same
    channel sequence; MAHO
Index Terms: FREQUENCY CHANGE METHOD ADJACENT CELL TDMA TELECOMMUNICATION
      VOICE  CHANNEL CHANGE NEW CARRY SEQUENCE SIGNAL CHANNEL TRANSFER
    CHANNEL SEQUENCE
Patent Assignee: (MOTI ) MOTOROLA INC
Author (Inventor): BORTH D E; HAUG J R; RASKY P D
Number of Patents: 005
Number of Countries: 005
Patent Family:
Patent No Kind  Date   Week Applic No    Date  LA  Pages IPC
FR 2696602  A1 940408 9417 FR 9311649  930930       26 H04B-007/26    (B)
FI 9304361  A  940403 9424 FI 934361   931004          H04B-007/26
SE 9303204  A  940403 9429 SE 933204   931001          H04J-003/00
US 5381443  A  950110 9508 US 955793   921002       14 H04L-027/30
IL 107058   A  960514 9633 IL 107058   930921          H04B-001/713
Priority Data (CC No Date): US 955793 (921002)
Applications (CC,No,Date): IL 107058 (930921); FR 9311649 (930930); FI
    934361 (931004); SE 933204 (931001)
Abstract (Basic): FR 2696602 A
        The signalling channel is divided into two blocks of line
    separated by N/2 line blocks in the N line blocks multiplexed in the
    TDMA frame. The first block contains synchronisation information and
    the second block contains information on the jump sequence.
        Access is just gained on one carrier, among a number, to which
    the signalling channel and many  voice  channels are jumped. When the
    signalling frequency is changed, synchronisation is obtained and
    information on the jump sequence is obtained. The signalling channel
    has  sub - channels  for signal broadcast, synchronisation,
    personal calling, short messages, acceptance of access and broadcast
```

data.

USE/ADVANTAGE - Mobile assisted hand-over in cellular communications. Communication quality maintained at cell boundary, simpler configuration. Possible lower cost version for slower operation.

Dwg.5/12

Abstract (US): 9508 US 5381443 A

The method of accessing a frequency hopping time division multiple access communication system involves accessing a carrier out of several carriers. The carriers have a signalling channel and a number of traffic channels frequency hopped thereon using a hopping sequence. Synchronisation to the signalling channel is acquired when the carrier has the signalling channel hopped on it. After synchronisation has been acquired, information related to the hopping sequence is obtained from the signalling channel.

Preferably, synchronisation to the signalling channel is acquired during a first timeslot of N timeslots multiplexed onto a TDMA frame. Information related to the hopping sequence is obtained during a second timeslot.

USE/ADVANTAGE - Cellular communications system. Supports logical channels for broadcast, paging , synchronisation, access and access grant. Simple configuration.

Dwg.1/12

File Segment: EPI
Derwent Class: W02;
Int Pat Class: H04B-001/713; H04B-007/212; H04B-007/26; H04J-003/00; H04L-027/30; H04Q-007/00
Manual Codes (EPI/S-X): W02-C03C1A; W02-C03C1D

36/9/2 (Item 2 from file: 351)
DIALOG(R)File 351:DERWENT WPI

004451009 WPI Acc No: 85-277887/45
XRPX Acc No: N85-207272

Offset type direct modulation FM data receiver has non-linear discriminator function producing voltage with sign to adjust local oscillator frequency

Index Terms: OFFSET TYPE DIRECT MODULATE FM DATA RECEIVE; NON LINEAR DISCRIMINATE FUNCTION PRODUCE VOLTAGE SIGN ADJUST LOCAL OSCILLATOR FREQUENCY
Patent Assignee: (PHIG) PHILIPS GLOEILAMPENFAB NV; (PHIG) PHILIPS ELECTRONIC
Author (Inventor): MARSHALL C B; WILSON J F; NETTLESHIP R
Number of Patents: 009
Number of Countries: 007
Patent Family:

Patent No	Kind	Date	Week	Applic No	Date	LA	Pages	IPC	
EP 160339	A	851106	8545			Eng	27		(B)
GB 2158330	A	851106	8545						
JP 60237749	A	851126	8602						
DK 8501885	A	851031	8606						
AU 8541780	A	860605	8630						
US 4672636	A	870609	8725						
EP 160339	B	920108	9203						
DE 3585098	G	920220	9209						
DK 165536	B	921207	9303	DK 851885	850426			H04L-027/14	

Priority Data (CC No Date): GB 8411037 (840430)

Applications (CC,No,Date): DK 851885 (850426)
Language: English
EP and/or WO Cited Patents: A3...8801; GB 1106913; GB 2059702; GB 2109201; GB 2122437; No-SR.Pub; US 4103244
Designated States
(Regional): DE; FR; GB
Filing Details: DK0165536 Previous Publ. DK 8501885
Abstract (Basic): EP 160339

The receiver has a mixer (14). The mixer has a first input receiving a directly modulated FM signal with two signalling frequencies deviated by delta F on either side of a carrier frequency and a second input. The second input receives local oscillator signal with a frequency between the two signalling frequencies but offset from the carriers frequency by a predetermined amount (df), which is smaller than the deviation delta f. A demodulator distinguishes between the signalling tones (delta f+df) and (delta f-df) and derives. an output signal.

A functional system has a frequency voltage transfer function which is non-linear when the receiver is tuned to the nominal frequency within the region occupied by the channel data signals. The functional system produces an output voltage of such a sign over the relevant frequency range as to tune the local oscillator frequency on to the desired offset frequency.

USE - For e.g. digital paging . @(27pp Dwg.No.1/22)@

Abstract (US): 8725 US 4672636

A subcircuit · in the AFC system has a symmetrical triangular voltage-frequency transfer function with a vertex of the function occurring at the deviation frequency and an output of the sub - circuit is coupled to a frequency control input of a local oscillator. The vertex may occur at a maximum or a minimum voltage. The transfer function enables the correct AFC output to be obtained for both of the signalling tones.

The AFC system may be implemented in an analogue or digital form. An analogue implementation of the sub - circuit having triangular transfer function, with a vertex at a maximum voltage, comprises a multiplier having one input coupled to the output of the mixer and another input connected to an output of a delay device whose input is coupled to the mixer. A smoothing circuit is connected to the output of the multiplier to remove the high frequency product term, also to define the time constant of the AFC system. @(13pp)@

Abstract (EP): 9203 EP 160339

A direct modulation FM data receiver comprising a mixer having a first input for receiving a directly modulated FM signal having two signalling input, alocal oscillator (20) for generating at an outputthereof an output having an output signal having a frequency which is offset from fc by a predetermined amount whicn is smaller than delta f, which output is coupled to said second input, demodulating means having an input coupled to the mixer output, for distinguishing between frequencies (delta f + df) and deriving an output data signal therefrom,and an AFC feedback path coupling the mixer (14) output to the frequency control input of the local oscillator (20), for reducing errors in the frequency of the local oscillator output signal, characterised in that the AFC feedback path has an output voltage versus input frequency charatcteristic which has slopes of opposite signs at frequencies of (delta f + f) and (delta f -df) respectively, and in whichthe AFC feedback path comprises a circuit (50 or 56) having first and second inputs to which the mixer output is coupled and an output (52) coupled to saic frequency control input, for effectively multiplying together signals presented to its inouts and producing a signal representat ive of the result at its output and delay means

included in the coupling from the mixer output to the first input of
said circuit. @(17pp)@
File Segment: EPI
Derwent Class: W01; W05; R57; R55; R56; R54
Int Pat Class: H03L-007/06; H04L-027/14; H03J-007/04; H04B-001/16
Manual Codes (EPI/S-X): W01-A09A; W05-A05
?

```
S17     10574  BASESTATION? OR LANDSTATION? OR (BASE OR LAND)()STATION?

S25    112785  DEMODULAT? OR DE()MODULAT? OR MODEM? OR S1(3N)DIGIT?

S37     45041  S25 AND (RECEIV? OR TERMINAL?)
S38      7303  S37 AND FILTER?
S39      2952  S38 AND (MODULAT? OR QAM)
S40      1348  S39 AND AMPLI?
S41       169  S40 AND SIGNAL?(2N)PROCESS?
S42         1  S41 AND S17
```

42/9/1 (Item 1 from file: 351)
DIALOG(R)File 351:DERWENT WPI

007160533 WPI Acc No: 87-157542/23
Related WPI Accession(s): 87-157542
XRPX Acc No: N87-118199
Subscriber equipment for wireless digital telephony - has control and MODEM processors directly coupled in assembly containing CODEC and universal transmitter receiver
Index Terms: SUBSCRIBER EQUIPMENT WIRELESS DIGITAL TELEPHONE CONTROL MODEM PROCESSOR COUPLE ASSEMBLE CONTAIN CODEC UNIVERSAL TRANSMIT RECEIVE
Patent Assignee: (ITMO) INT MOBILE MACHINES CORP; (ITMO-) INT MOBILE MACH
Author (Inventor): CRITCHLOW D N; PANETH E; SMETANA B A; YEHUSHUA M; AVIS G M; EARLAM S J K; JOHNSON K J; WESTLING G L; ERLAM S J K; ISRAEL G; AVIS G N
Number of Patents: 095
Number of Countries: 022
Patent Family:

Patent No	Kind	Date	Week	Applic No	Date	LA	Pages	IPC
BE 905822	A	870527	8723	BE 905822	861127		35	(B)
DE 3644066	A	880211	8807	DE 3644066	861222			
GB 2194403	A	880302	8809	GB 8717221	870721			
GB 2194404	A	880302	8809	GB 8717218	870721			
GB 2194417	A	880302	8809	GB 8717217	870721			
GB 2194418	A	880302	8809	GB 8717219	870721			
GB 2194708	A	880309	8810	GB 8717220	870721			
GB 2194711	A	880309	8810	GB 8627428	861117			
NL 8700645	A	880301	8813					
SE 8604661	A	880208	8813					
FR 2602622	A	880212	8814					
JP 63043430	A	880224	8814	JP 8774573	870330			
NO 8604618	A	880229	8814					
FR 2602928	A	880219	8815					
FR 2602929	A	880219	8815					
FR 2602933	A	880219	8815					
FR 2602934	A	880219	8815					
FR 2602935	A	880219	8815					
FR 2602938	A	880219	8815					
FR 2602941	A	880219	8815					
FR 2602944	A	880219	8815					
BR 8701441	A	880322	8817					
DK 8701789	A	880208	8818					
FI 8604943	A	880208	8818					
AU 8781363	A	880317	8819					
AU 8781364	A	880317	8819					
AU 8781365	A	880317	8819					
AU 8781366	A	880317	8819					

AU 8781367	A	880317	8819						
AU 8781368	A	880317	8819						
AU 8781369	A	880317	8819						
AU 8781370	A	880317	8819						
AU 8781371	A	880317	8819						
GB 2198915	A	880622	8825	GB 8717222	870721				
GB 2198916	A	880622	8825	GB 8717223	870721				
GB 2199206	A	880629	8826	GB 8717225	870721				
GB 2199214	A	880629	8826	GB 8717224	870721				
FR 2607336	A	880527	8828						
PT 85503	A	880817	8838						
US 4825448	A	890425	8919	US 893916	860807		12		
ES 2003951	A	881201	8933	ES 3224	861128				
CA 1263900	A	891212	9003						
US 4881240	A	891114	9004	US 256557	881012		12		
CN 1031306	A	890222	9007						
US 4893317	A	900109	9010	US 256415	881012		12		
FI 9000737	A	900214	9022						
CH 674435	A	900531	9024						
US 4943983	A	900724	9032	US 256580	881012				
CA 1272316	B	900731	9036						
CA 1272317	B	900731	9036						
CA 1272817	A	900814	9040						
CA 1274630	A	900925	9045						(NC)
GB 2194708	B	901121	9048						
GB 2194404	B	910306	9111						(NC)
GB 2199206	B	910313	9111						
US 4994802	A	910219	9111	US 256579	881012				(NC)
US 4996697	A	910226	9111						
GB 2194403	B	910327	9114						(NC)
GB 2194417	B	910424	9118						(NC)
GB 2194418	B	910424	9118						(NC)
GB 2194711	B	910424	9118						(NC)
GB 2198915	B	910424	9118						(NC)
GB 2198916	B	910424	9118						(NC)
GB 2199214	B	910501	9119						(NC)
NL 9002797	A	910402	9119						(NC)
NL 9002798	A	910402	9119						(NC)
NL 9002799	A	910402	9119						(NC)
NL 9002800	A	910402	9119						(NC)
NL 9002801	A	910402	9119						(NC)
NL 9002802	A	910402	9119						(NC)
NL 9002803	A	910402	9119						(NC)
NL 9002805	A	910402	9119						(NC)
NL 9002806	A	910402	9119						(NC)
IT 1207336	B	890517	9132						(NC)
KR 9005142	B	900720	9133						(NC)
IL 80497	A	910730	9136						(NC)
IL 94835	A	910730	9136						(NC)
IL 94836	A	910730	9136						(NC)
IL 94837	A	910730	9136						(NC)
IL 94838	A	910730	9136						(NC)
IL 94839	A	910730	9136						(NC)
US 5067141	A	911119	9150						(NC)
US 5101418	A	920331	9216				11		
CA 1303686	C	920616	9230	CA 534177	870408	Fre		H04M-001/00	
				CA 608564	890816				
CA 1303687	C	920616	9230	CA 534177	870408	Fre		H03D-007/00	
				CA 608563	890816				
US 5159705	A	921027	9246	US 893916	860807		13	H04B-001/40	

				US 256578	881012	
				US 578332	900906	
				US 853334	920313	
US 5168507	A	921201	9251	US 893916	860807	19 H03H-007/30
				US 256557	881012	
				US 433995	891109	
US 5177741	A	930105	9304	US 893916	860807	12 H04J-003/12
				US 256577	881012	
NO 172090	B	930222	9313	NO 864618	861119	H04M-003/42
CA 1319957	C	930706	9333	CA 534177	870408	H04B-001/40
				CA 608565	890816	
IE 67260	B	960306	9625	IE 863045	861118	H04L-027/20
IE 67262	B	960306	9625	IE 863045	861118	H04L-027/20
				IE 913835	861118	
IE 67263	B	960306	9625	IE 863045	861118	H04L-027/20
				IE 913836	861118	
IE 67264	B	960306	9625	IE 863045	861118	H04Q-007/00
				IE 913837	861118	
IE 67265	B	960306	9625	IE 863045	861118	H04L-027/00
				IE 913838	861118	

Priority Data (CC No Date): US 893916 (860807); US 256415 (881012); US 256416 (881012); US 256580 (881012); US 256579 (881012); US 256578 (881012); US 578332 (900906); US 853334 (920313); US 433995 (891109); US 256577 (881012)

Applications (CC,No,Date): IE 863045 (861118); BE 905822 (861127); DE 3644066 (861222); GB 8717221 (870721); GB 8717218 (870721); GB 8717217 (870721); GB 8717219 (870721); GB 8717220 (870721); GB 8627428 (861117); JP 8774573 (870330); GB 8717222 (870721); GB 8717223 (870721); GB 8717225 (870721); GB 8717224 (870721); ES 3224 (861128); US 256557 (881012); CA 534177 (870408); CA 608564 (890816); CA 534177 (870408); CA 608563 (890816); US 893916 (860807); US 893916 (860807); US 256557 (881012); US 893916 (860807); NO 864618 (861119); CA 534177 (870408); CA 608565 (890816); IE 863045 (861118); IE 913835 (861118); IE 863045 (861118); IE 913836 (861118); IE 863045 (861118); IE 913837 (861118); IE 863045 (861118); IE 913838 (861118)

Language: French

Filing Details: US5159705 Div ex US 4825448; US5168507 Div ex US 4825448; US5168507 Cont of US 4881240; US5177741 Div ex US 4825448; NO0172090 Previous Publ. NO 8604618

Abstract (Basic): BE 905822 A

A base-band processor provides control of conversion of PCM signals at one bit rate to other bit rates. It converts received digital signals to voice signals and vice versa. It also provides echo cancellation where it uses dynamic memory to store received signals. PROM's store the echo cancellation program information as well as that for utilisation of the processor as a control processor.

As a control processor it signals to the frequency synthesiser the frequency of operation for communication to the base station. The processor is directly coupled to a modem processor which is able to access the base band processor memory. This processor sends its signal at a predetermined sampling rate which are converted to analogue signals. The analogue signals are subjected to a cancellation process to reduce distortion. These signals are converted to an IF for addition to the synthesiser frequency to result in an HF signal for transmission.

ADVANTAGE - Combined assembly of modem and baseband processors. Dwg.0/10

Abstract (US): 9304 US 5177741 A

In the subscriber unit a selector determines the type of channel in an incoming signal and the unit transmit or receive mode . The selector produces repetitive frames at predetermined intervals where a portion of each interval constitutes an Amplitude Modulation Hole, determines whether the unit is in a transmit or receive mode and whether a particular channel is a control channel or a voice channel in accordance with the AM hole duration.

A portion of the first half of each frame constitutes a receive mode and a portion of the second half of each frame constitutes a transmit mode, each portion comprising a slot, each slot containing, as part of its initial data, a unique word establishing timing for reception of the remaining data in the slot.

Dwg.1/10 9251 US 5168507 A

The digital wireless subscriber telephone unit includes an IF used within the subscriber unit and base station which is combined with a carrier signal generated by a synthesiser for RF transmission between stations. In intervals between actuation of the system, a training mode is used. A known training signal is compared with a known value to produce correction coefficients to compensate for undesirable variations in timing and-or frequency. The constants are stored for use in correcting actual received signals.

During demodulation , the modulated digital signals are fed to the modem processor in the form of time multiplexed In-phase and (Quadrature) samples and are demultiplexed. The demultiplexed I and Q samples are fed to an equaliser and frequency correction circuit for minimisation of errors for production of a frequency correction signal based upon the correction coefficients which is used to correct any errors in the timing and frequency of the system.

USE/ADVANTAGE - Subscriber system adapted for communication with base station , direct access to processors and memories for easy maintainence and test purposesDwg.8/10 9246 US 5159705 A

The frequency synthesizer circuit comprises an oscillator that outputs a synchronising signal, and a control input signal generator. The synthesiser unit receives the synchronising signal, and the control input signal. The synthesiser unit, in response to the control input signal, generates an output signal which is in synchronism with the synchronising signal.

A frequency divider receives the synchronising signal, divides it by a predetermined value and outputs a divided synchronising signal. A mixer mixes the synthesiser output signal with the divided synchronising signal and generates a second synthesiser output signal which is in synchronism with the first output signal, but has a frequency which is lower than the first output signal by the value of the divided synchronising signal.

USE/ADVANTAGE - Subscriber system unit for digital radio telephone system, transcoding incoming and outgoing signals between bit stream types, provides echo cancellation.

Dwg.6/10 9216 US 5101418 A

The digital time-multiplexed quadrature frequency upconverter has input for receiving a first predetermined timing signal and input device for receiving a time-multiplexed quadrature signal. A signal combining device performs a multiplication function in combining the set timing signal with the time-multiplexed quadrature signal.

A digital-to-analog converter converts the combined signal to a first analog signal. The second signal combining device for the first analog signal output is provided from the digital -to- analog converter with a second analog signal at a set frequency to generate a third analog signal. A device integrates the third analog signal to generate the second signal centred at a second frequency.

USE - For converting signal centred at initial frequency to second signal centréd at second initial frequency9111 US 4994802

The system incldues a circuit to produce an analog signal, and a circuit for amplifying this signal, mixing the amplified signal with a signal from a timing circuit and applying the output of the mixer to a sample and hold circuit. The output of the sample and hold circuit is then applied to an analog to digital converter, the output of which is applied to a FIFO, whose output is then available for further processing.

The functions of many of the elements e.g. a full speed PROM (44), the FIFO (16) an interpolator (48) and a PAL (50) may be provided within a modem processor of sufficiently large capacity. This may also be true of such elements as frame timing (91), blanking generation (58) the timer (51) divide-by-4, the divide-by-5 and some or all of a synthesizer (72). @(12pp)@ 9032 US 4943983

The modem has a processor including a DPSK converter coupled to a filter . The converter has a digital bit input and has an inverse Gray coding function whose output is applied to a phase quantiser for determining the absolute value of the current symbol. The quantiser is coupled to a differential encoder for providing a differentially encoded phase value that represents the modulo sum of the current differential phase and the prior absolute phase. The modulo sum is directly applied to form the I and Q of the current symbol.

The filter is adapted to create an oversampled PSK waveform from the I and Q components that provide a time division multiplexed signal.

USE/ADVANTAGE - For subscriber systems unit for digital remote telephone system. Minimal errors. @(12pp)@ 9010 US 4893317

A signal from a modem processor is converted to an analog signal, which is subjected to deglitching, and then upconverted and filtered to form an IF signal for amplification . The amplified IF signal is combined with a signal generated by a synthesiser to provide an RF signal. On the basis of certain signals received from the base station , the baseband processor produces initiating signal which determine whether the subscriber unit will be in the transmit or the receive mode.

In intervals between actuation of the system, a training mode is used in which a known signal from the modem processor is compared with a looped-back signal to produce correction constants to compensate for undesirable variations in the IF, these constants being stored for use in correcting actual received signals. During demodulation , the modulated digital signals are fed to the modem processor in the form of time multiplexed I and Q samples and are demultiplexed, equatised and frequency corrected to minimise errors. @(12pp)@ 9004 US 4881240

The digital wireless subscriber telephone unit for wireless connection with a base station has a baseband processor which is coupled to storage device for the baseband functions. The baseband processor is direct access coupled to a modem processor whereby the two processors may communicate with each other. The modem processor acts as the master in the system, although lock-out circuit may be used in certain circumstances. The signal from the modem processor is converted to an analog signal, which is subjected to deglitching, and the deglitched signal is then upconverted and filtered to form an IF signal which is thereafter amplified . The amplified IF signal is combined with a signal generated by a synthesizer to provide an RF signal.

On the basis of certain signals received from the base station , the baseband processor produces initiating signals which determine whether the subscriber unit will be in the transmit mode or the receive mode. In intervals between actuation of the system, a training mode is used wherein a known signal from the modem processor is compared with a looped-back signal to produce correction constants to compensate for undesirable variations in the IF, these constants being stored for use in correcting actual received signals. @(12pp)@ 8919 US 4825448

The digital wireless subscriber telephone unit for wireless connection with a base station . The subscriber unit has baseband processor which is coupled to storage means for the baseband functions. The baseband processor is direct access coupled to a modem processor whereby the two processors may communicate with each other. The modem processor generally acts as the master in the system, although lock-out means may be used in certain circumstances. The signal from the modem processor is converted to an analog signal, which is subjected to deglitching, and the deglitched signal is then upconverted and filtered to form an IF signal which is thereafter amplified . The amplified IF signal is combined with a signal generated by a synthesiser to provide an RF signal. On the basis of certain signals received from the base station , the baseband processor produces initiating signals which determine whether the subscriber unit will be in the transmit mode or the receive mode.

In intervals between actuation of the system, a training mode is used wherein a known signal from the modem processor is compared with a looped-back signal to produce correction constants to compensate for undesirable variations in the IF, these constants being stored for use in correcting actual received signals. @(12pp)@

Abstract (GB): 9120 GB 2194418

An interpolator for varying the sampling rate of a digital signal having time division multiplexed I and Q components at an initial freq., comprising an input for said signal; a memory for the I or Q component serially coupled to a memory for the Q or I component; and means for applying a clock input of a predetermined freq. to both of said memories, whereby said time division multiplexed I and Q components are recirculated through the memories, during which they are demultiplexed at their initial freq. and then resampled and remultiplexed at a second higher frequency 9119 GB 2199214

A digital time-multiplexed quadrature frequency upconverter for converting a first time-multiplexed quadrature signal centred at an initial frequency to a second signal centred at a second frequency comprising an input means for said first signal; an input means for a carrier signal; a multiplier for acting on said first signal and for performing a time-multiplexed quadrature mixing thereof with the carrier signal; a digital -to- analog converter for converting said multiplied signal to an analog signal; and means for integrating the resultant analog signal. GB 2198916

A symbol timing tracking and AFC system comprising an input for a detected phase signal; means for subtracting a phase correction value from the detected phase signal to provide a corrected phase signal; means for quantizing the phase of the corrected phase signal to a predetermined increment; means for subtracting the quantized phase signal from the corrected phase signal to provide a phase error signal; and means for obtaining said phase error signal and computing both a phase correction value and a freq. correction signal therefrom

File Segment: EPI

Derwent Class: W01;

Int Pat Class: G11C-027/02; H01M-001/00; H03B-021/02; H03C-003/06; H03D-007/00; H03H-007/06; H03H-007/30; H03K-005/15; H03L-007/16; H03M-001/06; H03M-007/30; H04B-001/40; H04B-007/21; H04B-007/24; H04B-007/26; H04J-001/05; H04J-003/02; H04J-003/06; H04J-003/12;

H04J-011/00; H04L-005/12; H04L-005/22; H04L-007/02; H04L-011/00;
H04L-025/03; H04L-027/18; H04L-027/34; H04M-001/00; H04M-003/42;
H04Q-007/04; H04Q-011/04
Manual Codes (EPI/S-X): W01-B05; W01-C01D
?

```
Set     Items   Description
S1      227391  VOICE OR AUDIO OR ANALOG?

S25     112785  DEMODULAT? OR DE()MODULAT? OR MODEM? OR S1(3N)DIGIT?

S37      45041  S25 AND (RECEIV? OR TERMINAL?)
S38       7303  S37 AND FILTER?
S39       2952  S38 AND (MODULAT? OR QAM)
S40       1348  S39 AND AMPLI?
S41        169  S40 AND SIGNAL?(2N)PROCESS?

S43         44  S41 AND S1(3N)DIGIT?
S44          0  S43 AND TRANSFORM?(2N)FILTER?
S45         11  S43 AND (QUADRATURE? OR QAM)

S48         10  S45 NOT (S9 OR S11 OR S16 OR S20 OR S24 OR S27 OR S36 OR S-
             42)
?i
t48/9/all
```

48/9/1 (Item 1 from file: 351)
DIALOG(R)File 351:DERWENT WPI

010411730 WPI Acc No: 95-313044/41
XRPX Acc No: N95-236587 *Image available*
Digital *signal* synthesis and *processing* in full duplex data transmission system - providing digital phase continuous synthesis as well as digital correlative *signal* *processing* at reception end via current supply networks by means of narrow band or band spreading *modulation*
Index Terms: DIGITAL *SIGNAL* SYNTHESIS *PROCESS* FULL DUPLEX DATA TRANSMISSION SYSTEM DIGITAL PHASE CONTINUOUS SYNTHESIS WELL DIGITAL CORRELATE *SIGNAL* *PROCESS* RECEPTION END CURRENT SUPPLY NETWORK NARROW BAND BAND SPREAD *MODULATE*
Patent Assignee: (ALLM) ABB PATENT GMBH
Author (Inventor): DOSTERT K
Number of Patents: 002
Number of Countries: 006
Patent Family:

Patent No	Kind	Date	Week	Applic No	Date	LA	Pages	IPC	
EP 671822	A2	950913	9541	EP 95103147	950306	Ger	17	H04B-001/713	(B)
DE 4408334	A1	950914	9542	DE 4408334	940311		15	H04L-027/00	

Priority Data (CC No Date): DE 4408334 (940311)
Applications (CC,No,Date): EP 95103147 (950306)
Language: German
EP and/or WO Cited Patents: No-SR.Pub
Designated States
(Regional): AT; CH; DE; FR; LI; SE
Abstract (Basic): EP 671822 A
On reception, a *filtered* and *amplified* reception signal (X) coupled from the current network, reaches a digital multiplier (5) after *digital* to *analog* conversion. The second input of the multiplier has a reference signal (Y) from a sample value memory (24) addressed by a cyclic address counter to fixed base length (20). Exactly four reference values (Y), corresponding to the in-phase and *quadrature* sample values of each L and H signal form, are allocated to each X value. The resulting products are accumulated in a ring

structure of a summer (6), four registers (7,8,9,10) and a switch (11).

After one signal form cycle, the in-phase components of the H and L bit are accumulated in two registers (9,10) and the *quadrature* components are accumulated in the other registers (7,8). By undoing the ring structure, the switch (11) feeds the four results from the registers (7-10) to a decision unit including a switch (13), a summer (14) and two registers (15,16).

ADVANTAGE - Can transmit and *receive* while carrying out full duplex operation. Has high flexibility in data rate, interference resistance and frequency selection. Dwg.1/3

File Segment: EPI
Derwent Class: W01; W02; X12;
Int Pat Class: H04B-001/713; H04B-003/54; H04L-005/14; H04L-027/10
Manual Codes (EPI/S-X): W01-A03D; W02-C01A3; W02-K05A1; X12-H03E

48/9/2 (Item 2 from file: 351)
DIALOG(R)File 351:DERWENT WPI

010040810 WPI Acc No: 94-308521/38
XRPX Acc No: N94-242712 *Image available*
Orthogonal *modulator* e.g. QPSK for *signal* *processing* equipment - involves *quadrature* phase shift keying of signals *received* along different channels and inducing phase differences for distinguishing each signal
Index Terms: ORTHOGONAL *MODULATE* QPSK *SIGNAL* *PROCESS* EQUIPMENT *QUADRATURE* PHASE SHIFT KEY SIGNAL *RECEIVE* CHANNEL INDUCE PHASE DIFFER DISTINGUISH SIGNAL
Patent Assignee: (FUIT) FUJITSU LTD
Number of Patents: 001
Number of Countries: 001
Patent Family:

Patent No	Kind	Date	Week	Applic No	Date	LA	Pages	IPC	
JP 6237276	A	940823	9438	JP 9323102	930212		9	H04L-027/20	(B)

Priority Data (CC No Date): JP 9323102 (930212)
Abstract (Basic): JP 06237276 A

The orthogonal *modulator* consists of two channels, each incorporating a digital *filter* (11, 21), multipliers (2, 3) and a multiplier (1) linked to them, *digital* to *analog* convertors (12, 22), wave *filters* (13, 23). 90deg. phase compensation circuits (15, 25), *amplitude* regulator (14, 24), signal dispatcher (30), local oscillator (30) and hybrid vessel (40).

Digital data from each of the two channels is multiplied with differential phase retentions using multipliers (1, 2, 3) and converted into *analog* signal using *digital* to *analog* convertor (12, 22). Higher harmonies that may produce distortion are removed in the wave *filter* (13, 23) and the signal level is adjusted for *amplitude* setting in the *amplitude* regulator (14, 24). Each signal is mixed with the carrier wave from the local oscillator, routed to different channels in the signal dispatcher and retrieved as a quadruple phase shift keying signal when combined in the hybrid vessel.

ADVANTAGE - Improves stability of signal generation.
Dwg.1/8

File Segment: EPI
Derwent Class: U23; W01;
Int Pat Class: H04L-027/20
Manual Codes (EPI/S-X): U23-P01; W01-A09B; W01-A09E1

48/9/3 (Item 3 from file: 351)
DIALOG(R)File 351:DERWENT WPI

009521164 WPI Acc No: 93-214705/27
XRPX Acc No: N93-165018
Vector FM transmitter suitable for cellular telephones - uses sine and cosine circuitry for generating *quadrature* and in-phase *modulating* signals
Index Terms: VECTOR FM TRANSMIT SUIT CELLULAR TELEPHONE SINE COSINE CIRCUIT GENERATE *QUADRATURE* PHASE *MODULATE* SIGNAL
Patent Assignee: (NOVA-) NOVATEL COMMUNICATIONS LTD
Author (Inventor): BARABASH D W; SMIT T J
Number of Patents: 002
Number of Countries: 002
Patent Family:

Patent No	Kind	Date	Week	Applic No	Date	LA	Pages	IPC	
CA 2080786	A	930418	9327	CA 2080786	921016		12	H03C-003/00	(B)
US 5224119	A	930629	9327	US 778567	911017		6	H04L-027/20	

Priority Data (CC No Date): US 778567 (911017)
Applications (CC,No,Date): CA 2080786 (921016)
Abstract (Basic): CA 2080786 A

Speech signals from a microphone (12) are processed in accordance with one of alternative processing sequences before being applied to a common vector *modulator* (14) which can include a *digital*-to-*analog*-conversion capability by two converters (16, 18) *receiving* in-phase and *quadrature* *modulating* signals, respectively, and applying them to respective multipliers (20, 22).

A phase splitter (24) generates in-phase and *quadrature* versions of a carrier signal, which are applied as the other inputs to the multipliers respectively. An analog summing circuit (26) adds the resultant product signals to generate an output which an *amplifier* (28) applies to an antenna (30) for transmission.

The vector *modulator* produces the desired FM signal by adding together *quadrature* and in-phase versions of a carrier in proportion to the values that the *quadrature* and in-phase *modulating* signals represent.

ADVANTAGE - Same vector *modulator* can be used for both frequency and phase *modulation* modes. Integration and computation of trigonometric functions can readily be performed by same digital-*signal*-*processing* circuitry that would be used for the various encoding and *filtering* steps required for digital phase-shift-keying standard. Allows implementation that does not require individual adjustments ordinarily associated with V.C.O. frequency-*modulation* circuitry.

Dwg.1/3

File Segment: EPI
Derwent Class: U23; W01; W02;
Int Pat Class: H03C-003/00; H04B-001/04; H04L-027/20
Manual Codes (EPI/S-X): U23-H; W01-C01D3A; W02-C03C1C; W02-G01D

48/9/4 (Item 4 from file: 351)
DIALOG(R)File 351:DERWENT WPI

009488425 WPI Acc No: 93-181960/22

XRPX Acc No: N93-139911 *Image available*
Partial response *demodulation* *digital* *audio* communication - includes 30 channel digital bit stream demultiplexed into nx groups of five channels
Index Terms: RESPOND *DEMODULATE* *DIGITAL* *AUDIO* COMMUNICATE CHANNEL *DIGITAL* BIT STREAM GROUP FIVE CHANNEL
Patent Assignee: (SCAT) SCIENTIFIC ATLANTA INC
Author (Inventor): MONTREUIL L
Number of Patents: 003
Number of Countries: 029
Patent Family:

Patent No	Kind	Date	Week	Applic No	Date	LA	Pages	IPC	
US 5214390	A	930525	9322	US 851445	920316		21	H03D-003/00	(B)
WO 9319520	A1	930930	9340	WO 93US2240	930316	Eng	36	H03D-003/00	
AU 9338040	A	931021	9407	AU 9338040	930316			H03D-003/00	

Priority Data (CC No Date): US 851445 (920316)
Applications (CC,No,Date): AU 9338040 (930316); WO 93US2240 (930316)
Language: English
EP and/or WO Cited Patents: US 4886395
Designated States
(National): AU; BR; CA; CZ; FI; HU; JP; KR; NO; PL; RO; SK
(Regional): AT; BE; CH; DE; DK; ES; FR; GB; GR; IE; IT; LU; MC; NL; PT; SE
Filing Details: AU9338040 Based on WO 9319520
Abstract (Basic): US 5214390 A

The appts. includes a group of channels each *modulating* a carrier by a *quadrature* partial response (QPR) *process*. The QPR *signal*, an *amplitude* *modulated*, double sideband, carrier suppressed (AM DSBSC) signal, is then transmitted over the cable system to a multiplicity of subscribers, each of which has a QPR *demodulator* . The *demodulators* are of the decision feedback type having a modified Costas loop carrier recovery circuit. The grouping of an odd number (five) of *digital* *audio* channels per QPR mopdulator minimises error propagation due to the correlative QPR *demodulation* process. The decision feedback decoding is implemented in simple current nodes where a bilevel output from the decoded data is subtracted from a tertiary level output of a mixer.

The resulting bilevel current is converted to a voltage, *filtered*, and limited to produce a logic output level. The modified Costas loop provides an error voltage based upon the difference of the *received* data states versus the nominal data stages and the quadrant the data occupies. The error voltage is generated by differencing the outputs of two analog switches which have as their inputs analog signals representative of the *amplitude* level of the data and its inverse of one phase. The signal of the data bit of the opposite phase is used to control the switch and thus select which signal is used.

ADVANTAGE - Very efficient conversion.

Dwg.10/15

File Segment: EPI
Derwent Class: U23; W01; W04;
Int Pat Class: H03D-003/00
Manual Codes (EPI/S-X): U23-P04; W01-A02; W01-A09E2; W04-C10A1; W04-K05

48/9/5 (Item 5 from file: 351)
DIALOG(R)File 351:DERWENT WPI

009452023 WPI Acc No: 93-145548/18
XRPX Acc No: N93-111199 *Image available*

Digital *quadrature* radio *receiver* with two=step processing - uses *analogue*=to=*digital* converter followed by concurrent *filter* providing *quadrature* mixing and downsampling functions; DSP
Index Terms: DIGITAL *QUADRATURE* RADIO *RECEIVE* TWO-STEP PROCESS *ANALOGUE*-*DIGITAL* CONVERTER FOLLOW CONCURRENT *FILTER* *QUADRATURE* MIX FUNCTION
Patent Assignee: (FORD) FORD FRANCE SA; (FORD) FORD MOTOR CO LTD; (FORD) FORD WERKE AG; (FORD) FORD MOTOR CO
Author (Inventor): WHIKEHART W J; WHIKEHART J W
Number of Patents: 003
Number of Countries: 004
Patent Family:

Patent No	Kind	Date	Week	Applic No	Date	LA	Pages	IPC	
EP 540195	A2	930505	9318	EP 92309141	921005	Eng	16	H04B-001/16	(B)
US 5222144	A	930622	9326	US 783470	911028		14	H04H-005/00	
EP 540195	A3	940511	9524	EP 92309141	921005			H04B-001/16	

Priority Data (CC No Date): US 783470 (911028)
Applications (CC,No,Date): EP 92309141 (921005); EP 92309141 (921005)
Language: English
EP and/or WO Cited Patents: No-SR.Pub; 1.Jnl.Ref; EP 448150 A; US 4468794 A; US 4592074 D
Designated States
(Regional): DE; FR; GB

Abstract (Basic): EP 540195 A

The two-step digital *signal* *processing* (DSP) radio *receiver* includes a conventional analogue RF tuner to produce an analogue intermediate frequency (IF). The *receiver* includes an *analogue*-to-*digital* converter (40) which performs DSP functions of *digitally* sampling the *analogue* IF. A concurrent *filter* (41) provides *quadrature* mixing, resulting in an IF centre frequency near zero, and downsampling functions.

A *quadrature* mixer (42) is phase locked to provide synchronous detection by means of a loop *filter* (43) and a digital voltage-controlled oscillator (VCO) (44). During *demodulation* of a C-QUAM signal the *quadrature* mixer becomes locked to the near-zero Hz complex IF signal and produces at its outputs synchronous detector I and Q signals which together form a zero Hz IF signal.

USE/ADVANTAGE - for reception of Compatible-*Quadrature* *Amplitude* *Modulation* signals. Provides advantages of synchronous detection and low sampling rate simultaneously in digital *receiver*. Reduced DSP processing load.

Dwg.6/24

Abstract (US): 9326 US 5222144 A

The digital *signal* *processing* (DSP) radio *receiver* employs a conventional analog RF tuner to produce an analog inetrmediate frequency. The *receiver* performs DSP functions of *digitally* sampling the *analog* intermediate frequency at a sampling rate fs, concurrently mixing, *filtering*, and sample-rate reducing the sampled intermediate frequency using a digital *filter* for inherently generating a near-zero complex intermediate signal and synchronously detecting the complex IF signal.

USE - AM stereo signal reproduction.

Dwg.6/24

File Segment: EPI
Derwent Class: U23; W01; W03;
Int Pat Class: H04B-001/16; H04H-005/00
Manual Codes (EPI/S-X): U23-P01; U23-P04; W01-A09C; W01-A09E2; W03-B02C5

48/9/6 (Item 6 from file: 351)
DIALOG(R)File 351:DERWENT WPI

008373621 WPI Acc No: 90-260622/34
XRPX Acc No: N90-201905 *Image available*
Digital carrier *demodulator* having components working beyond limits - HAS A-D converter to *receive* if signal and output digital sample stream, two half-band LPFS and decimators
Index Terms: DIGITAL CARRY *DEMODULATE* COMPONENT WORK LIMIT; *ANALOGUE*-*DIGITAL* CONVERTER *RECEIVE* SIGNAL OUTPUT DIGITAL SAMPLE STREAM TWO HALF BAND
Patent Assignee: (USAS) NAT AERO & SPACE ADMIN
Author (Inventor): SADR R; HURD W J
Number of Patents: 001
Patent Family:

CC Number	Kind	Date	Week	
US 4947408	A	900807	9034	(Basic)

Priority Data (CC No Date): US 350813 (890512)
Filing Details: US4947408 (1810RMC)
Abstract (Basic): US 4947408
The device has *analog*-to-*digital* converter for *receiving* an intermediate frequency analog signal and for outputting a digital representation of it. A switch circuit samples the digital representation of the intermediate frequency analog signal from the A-D converter and for outputting a pair of digital sample streams. A multiplier multiplies one of the pair of digital sample streams with an in-phase carrier signal. A second multiplier multiplies the other of the pair of digital sample streams with a *quadrature* carrier signal. A half-band digital low-pass *filter* is provided for *filtering* an output from the first multiplier. A second half-band digital low-pass *filter* *filters* an output from the second multiplier.
A decimator is provided for decimating the number of samples from an output from the first half-band digital low-pass *filter*. The decimators each have an output which are input to a baseband *signal* *processor*.
USE - In reception of phase and *amplitude* *modulated* digital signals of bandwidths up to 15MHz. @(6pp Dwg.No.3/3)@
File Segment: EPI
Derwent Class: U23; R54;
Int Pat Class: H03D-001/06
Manual Codes (EPI/S-X): U23-K

48/9/7 (Item 7 from file: 351)
DIALOG(R)File 351:DERWENT WPI

007386726 WPI Acc No: 88-020661/03
Related WPI Accession(s): 87-093595
Analog-to-*digital* converter for adaptive television deghosting system - has IIR *filter* responsive to in-phase and *quadrature*-phase components of *modulated* TV signals; INFINITE IMPULSE RESPOND
Index Terms: *ANALOGUE* *DIGITAL* CONVERTER ADAPT TELEVISION SYSTEM *FILTER* RESPOND PHASE *QUADRATURE* PHASE COMPONENT *MODULATE* TELEVISION SIGNAL
Patent Assignee: (RADC) RCA CORP; (RADC) RCA THOMSON LICENSING CORP
Author (Inventor): LEWIS H G; SHEAU-BAO N; NG S B
Number of Patents: 003
Number of Countries: 004

Patent Family:

Patent No	Kind	Date	Week	Applic No	Date	LA	Pages	IPC
US 4686570	A	870811	8803	US 813255	851224		14	(B)
EP 228260	B1	930310	9310	EP 86309940	861218	Eng	10	H04N-005/21
DE 3687961	G	930415	9316	DE 3687961	861218			H04N-005/21
				EP 86309940	861218			

Priority Data (CC No Date): US 813255 (851224); US 824665 (860131)
Applications (CC,No,Date): DE 3687961 (861218); EP 86309940 (861218); EP 86309940 (861218)
Language: English
EP and/or WO Cited Patents: GB 2130839; GB 2146203
Designated States
(Regional): DE; FR; GB
Filing Details: DE3687961 Based on EP 228260
Abstract (Basic): US 4686570 A

The digital samples developing appts comprises an input *terminal* for applying the input signals and a unit for developing a clock signal having a predetermined frequency. The clock signal has two phases that are mutually *quadrature* phase related. A unit, coupled to the input *terminal* and to the clock *signal* developer, *processes* the input *signals* to produce *processed* *signals* including a further carrier signal *modulated* by the base-base signals. The further carrier signal is locked in frequency and phase to one of the phases of the clock signal.

An *analog*-to-*digital* converter is coupled to the input *signal* *processor* and responds to the phases of the clock signal for alternately sampling the *processed* *signals* at instants determined by the phases of the clock signal, respectively, to develop digital samples which alternately represent components of the *processed* *signals* that are respectively in-phase with and *quadrature* phase related to the further carrier signal.

ADVANTAGE - Cancels ghost signal components of television signal.
Dwg.0/9

Abstract (EP): 9310 EP 228260 B

A method of de-ghosting a video signal by means of signal offset circuitry, the circuitry comprising: a source (10-14) for providing an analog video signal having a training signal component; an *analog*-to-*digital* converter (26) coupled to said source for developing a digital signal representative of said analog video signal; first means (16, 18) coupled to said source for developing a control signal (T) corresponding to a predetermined interval; and second means (20-24) coupled to said *analog*-to-*digital* converter and responsive to said control signal for changing the correspondence between said analog video signals and said digital signal during said predetermined interval, characterized in that the method comprises selecting a portion of the vertical synchronizing interval of analog video signal to represent the training signal component, said portion having a dynamic range normally extending from black level to sync. tip level and which may include a ghost component which, when present, may tend to exceed the input signal dynamic range of said *analog* to *digital* converter (26) thus resulting in ghost *amplitude* clipping and a corresponding loss of ghost signal *amplitude* information; producing, by said first means, said control signal only during said selected portion of said vertical synchronizing interval of said analog video signal; altering said correspondence, with said second means (20,22,24), between said training signal level and said input signal range of said converter such that said *analog* to *digital* converter does not clip peaks of the ghost component of the training signal for preventing said loss of said ghost *amplitude* information; and

employing digital de-ghosting circuitry (28) connected to an output of said *analog* to *digital* converter (26) and responsive to said training signal having preserved ghost *amplitude* information for removing said ghost component from the active video portions of the digitized video signal provided by said *analog* to *digital* converter.
Dwg.2/2
File Segment: EPI
Derwent Class: U21; U23; W02; W03;
Int Pat Class: H04N-005/21; H04N-005/40
Manual Codes (EPI/S-X): U21-A03F; U23-P03; U23-P04; W02-G03B9; W03-A03

48/9/8 (Item 8 from file: 351)
DIALOG(R)File 351:DERWENT WPI

004763768 WPI Acc No: 86-267109/41
XRPX Acc No: N86-199667
Digital zero IF circuit for use in radio or radar *receivers* has generator providing two clock pulse trains in *quadrature* phase relationship for operating samplers in two signal paths; PAGE
Index Terms: DIGITAL ZERO CIRCUIT RADIO RADAR *RECEIVE*; GENERATOR TWO CLOCK PULSE TRAIN *QUADRATURE* PHASE RELATED OPERATE SAMPLE TWO SIGNAL PATH
Patent Assignee: (STTE) STC PLC; (INTT) INT STAND ELEC CORP
Author (Inventor): WONG A C C; ROLLEY R
Number of Patents: 006
Patent Family:

CC Number	Kind	Date	Week	
GB 2173364	A	861008	8641	(Basic)
EP 197708	A	861015	8642	
AU 8655212	A	861009	8647	
JP 61236204	A	861021	8701	
GB 2173364	B	881005	8840	
CA 1261924	A	890926	8945	

Priority Data (CC No Date): GB 858714 (850403)
Applications (CC,No,Date): EP 86302239 (860326); JP 8675636 (860403)
Language: English
EP and/or WO Cited Patents: A3...8802; DE 3138464; EP 80014; GB 1517121
Designated States
(Regional): AT; BE; CH; DE; FR; IT; LI; LU; NL; SE
Abstract (Basic): GB 2173364
A *modulated* r.f. signal is *received* and *amplified* (10). The *amplifier* r.f. signal then fed to two identical signal paths (11I, 11Q) where the signal is sampled and *digitised* by an *analogue*-to-*digital* converter (12I,12Q). The digitised signal is then passed through a digital *filter* arrangement (13I,13Q). Sampling of the analogue signal is controlled by two clock pulse trains in *quadrature* running at a frequency 2fc, where fc is the carrier frequency of the input signal.

The base-band content of the input signal is extracted in the form of two phase *quadrature* components (I and Q). Because all the *signal* *processing* is performed in digital circuitry the balance between the I and Q channels is maintained. *Quadrature* orthogonality is maintained throughout a wide range of frequencies fc, being a function of the clock pulse orthogonality which is relatively easy to achieve. There is also a high uniformity of channel bandwidths. @(4pp Dwg. No.1/2)@
Abstract (GB): 8840 GB 2173364

A digital zero IF circuit having first and second signal paths to which a radio frequency signal is applied, each path including a sampling and digitising means and a digital *filter* means, the circuit including a clock generator arranged to generate two clock pulse trains in *quadrature* phase relationship, each sampling means being operated in response to a respective one of the clock pulse trains.
File Segment: EPI
Derwent Class: W02; W05; R56; R54; R57; R19
Int Pat Class: H04B-001/30; H03D-003/00; H04L-027/22; G01S-007/30; G01S-013/50; H03D-007/00
Manual Codes (EPI/S-X): W02-G03X; W05-A05

48/9/9 (Item 9 from file: 351)
DIALOG(R)File 351:DERWENT WPI

004665908 WPI Acc No: 86-169250/26
XRPX Acc No: N86-126257
Frequency selective sampling detector for network analyser converts *analogue* signal to *digital* *signal* before heterodyning *process*
Index Terms: FREQUENCY SELECT SAMPLE DETECT NETWORK ANALYSE; CONVERT *ANALOGUE* SIGNAL *DIGITAL* *SIGNAL* HETERODYNE *PROCESS*
Patent Assignee: (HEWP) HEWLETT PACKARD CO; (YOKH) YOKOGAWA-HEWLETT PACKARD
Author (Inventor): HILTON H E
Number of Patents: 002
Patent Family:

CC Number	Kind	Date	Week	
US 4594555	A	860610	8626	(Basic)
JP 61116670	A	860604	8629	

Priority Data (CC No Date): US 665694 (841029)
Applications (CC,No,Date): JP 85243794 (851029)
Abstract (Basic): US 4594555

The appts. comprises a sample and hold device coupled to the bandpass *filter*, for sampling the carrier signal *received* from it at a seletable rate. A converter coupled to the sample and hold device converts the carrier signal *received* to digital data. A multiplier coupled to the converter multiplies the digital data *received* by a selectable signal to produce an in-phase component of the *modulation* of the carrier signal.

A second multiplier coupled to the converter multiplies the digital data *received* by a second selectable signal to produce the *quadrature* component of the *modulation* of the carrier signal. Two *filters* coupled to the two multiplier, respectively, *filters* undesired components of the *modulation* signals *received*.

ADVANTAGE - Since ADC is common to both in-phase and *quadrature* parts of the signal *amplification* and orthogonality are matched. Drift-free centre-frequency, bandwidth and phase reference detector. @(22pp Dwg.No.2/23)@
File Segment: EPI
Derwent Class: S01; U23; R54; R18; R56
Int Pat Class: H03D-005/00; G01R-023/17; H03D-007/00; H04B-001/26
Manual Codes (EPI/S-X): S01-D03C; S01-D05C; U23-J; U23-X

48/9/10 (Item 10 from file: 351)
DIALOG(R)File 351:DERWENT WPI

003593468 WPI Acc No: 83-E1666K/13
XRPX Acc No: N83-054454

Multimode radio *receiver* has two channels providing in-phase and *quadrature* mixed signals to and A-D converter followed by a digital *demodulator*

Index Terms: MULTIMODE RADIO *RECEIVE* TWO CHANNEL PHASE *QUADRATURE* MIX SIGNAL *ANALOGUE*-*DIGITAL* CONVERTER FOLLOW *DIGITAL* *DEMODULATE*
Patent Assignee: (INTT) INT STD ELEC CORP; (STTE) STC PLC
Author (Inventor): GOATCHER J K
Number of Patents: 007
Patent Family:

CC Number	Kind	Date	Week	
EP 74858	A	830323	8313	(Basic)
GB 2106734	A	830413	8315	
JP 58114532	A	830707	8333	
US 4470147	A	840904	8438	
GB 2106734	B	860115	8603	
EP 74858	B	870708	8727	
DE 3276738	G	870813	8733	

Priority Data (CC No Date): GB 8127797 (810915)
Applications (CC,No,Date): US 415968 (820908); EP 82304870 (820915)
Language: English
EP and/or WO Cited Patents: No.SR.Pub; US 4090145; GB 2040615
Designated States
(Regional): CH; DE; LI; SE

Abstract (Basic): The *receiver* includes two signal paths, in one of which a *received* signal is mixed with a local oscillator frequency running at the main transmission frequency and then *filtered* to give an in-phase signal. In the other path the *received* signal is mixed with the local oscillator frequency but with a relative phase shift and then *filtered* to provide a *quadrature* signal.

An *analogue*-to-*digital* converter stage is provided to digitise the signals and a digital *signal* *processor* is arranged to *demodulate* the digitised signals. This appts. is capable of *receiving* and *demodulating* two of the following types of *modulation*; phase, *amplitude*, frequency, and single sideband *modulation* and it can also be used in a single mode. (19pp)

Abstract (US): 8438 US 4470147

The *receiver* includes a signal path including circuitry for mixing the *received* signal with a local oscillator frequency running at the main transmission frequency which is the centre frequency of the side band. The mixed signal is *filtered* to provide a signal defined as an in-phase signal (I). A second signal path includes circuitry for mixing the *received* RF signal with the local oscillator frequency but with a relative phase shift. The second mixed signal is *filtered* to provide a second *filtered* signal defined as a *quadrature* signal Q.

The I and Q signals are converted into respective digitised I and Q signals, and a digital *signal* *processor* *demodulate* the digitised signals. @(9pp)@

Abstract (GB): 8603 GB 2106734

A radio *receiver* comprising a first signal path in which a *received* signal is mixed with a local oscillator frequency running at the same frequency (e.g. main transmission frequency) and then *filtered* to give a first mixed signal defined as an in-phase signal (I), a second signal path in which the *received* signal is mixed with the local oscillator frequency but with a substantially 90 deg.phase shift and then *filtered* to provide a second mixed signal defined as a *quadrature* signal Q; an *analogue* to *digital* converter stage to digitise the I and Q signals; and a digital *signal* *processor*

arranged to *demodulate* the digitised I and Q signals and including
means to derive *amplitude* (R) and/or phase (theta) of the
modulation signals from the digitised I and Q signals and means to
derive the respective output of any two or more of: a) phase
modulation, b) *amplitude* *modulation*, c) frequency *modulation*,
d) single side band *modulation*.

Abstract (EP): 8727 EP 74858

A radio *receiver* comprising a first signal path in which a *received* signal is mixed with a local oscillator and then *filtered* to give a first mixed signal defined as an in-phase signal I, a second signal path in which the *received* signal is mixed with the local oscillator frequency but with a substantially 90 deg. phase shift and then *filtered* to provide a second mixed signal defined as a *quadrature* signal Q, an *analogue* to *digital* converter stage to digitise and I and Q signals; and a digital *signal* *processor* arranged to *demodulate* the digitised I and Q signals and including means to derive *amplitude* R and phase theta of the *modulation* signals from the digitised I and Q signals, characterised in that the *receiver* is arranged to *receive* and *demodulate* at least single side band *modulated* (ssbm) signals, and in that there is means (L1) to subtract or add phase at the sideband offset frequency of the phase angle phi to provide a new phase angle theta, and means (M) to derive R sin theta or R Cos theta to produce digitised signal side band *demodulation* depending respectively on whether the phase is subtracted or added. @(11pp)@

File Segment: EPI

Derwent Class: U23; W02; R54; R56;

Int Pat Class: H03D-001/00; H03D-003/00; H03D-005/00; H04B-001/16

Manual Codes (EPI/S-X): U23-K; U23-L; W02-G03A; W02-G03X

?pause

>>> PAUSE started.

?

```
File   9:Business & Industry(R) Jul 1994-1996/Sep 10
         (c) 1996 Resp. DB Svcs.
File  15:ABI/INFORM(R) 1971-1996/Sep W2
         (c) 1996 UMI
File  16:IAC PROMT(R) 1972-1996/Sep 10
         (c) 1996 Information Access Co.
File  88:IAC BUSINESS A.R.T.S. 1976-1996/Sep W2
         (c) 1996 Information Access Co
File 148:IAC Trade & Industry Database 1976-1996/Sep 10
         (c) 1996 Info Access Co
File 275:IAC(SM) Computer Database(TM) 1983-1996/Sep 10
         (c) 1996 Info Access Co
File 621:IAC New Prod.Annou.(R) 1985-1996/Sep 10
         (c) 1996 Information Access Co
File 624:McGraw-Hill Publications 1985-1996/Sep 05
         (c) 1996 McGraw-Hill Co. Inc
File 636:IAC Newsletter DB(TM) 1987-1996/Sep 10
         (c) 1996 Information Access Co.
?ds

Set     Items   Description
S1     793853   VOICE OR AUDIO OR ANALOG?
S2      26348   S1(2N)SIGNAL?
S3    2261134   COMPRESS? OR REDUC? OR REDN?
S4       1922   SUBCHANNEL? OR SUBCIRCUIT? OR SUBFACILIT? OR SUBLINE? OR S-
             UBPATH? OR SUBLINK? OR SUB()(CHANNEL? OR CIRCUIT? OR FACILIT?
             OR LINE? OR PATH? OR LINK?)
S5    3842877   TIME
S6        597   SINGLE(1W)(SIDEBAND? OR SIDE()BAND? ?)
S7          7   S2(4N)S3 AND S4 AND S5
S8          0   S7 AND S6
S9         56   S1(4N)S3 AND S4 AND S5
S10         0   S9 AND S6

S34         4   RD S7 (unique items)
?
t34/6/all

 34/6/1     (Item 1 from file: 16)
05960064
TTC announces new GSM Option for the T-BERD 107A.
*FULL TEXT AVAILABLE IN FORMAT 7 OR 9* WORD COUNT:   327


 34/6/2     (Item 1 from file: 88)
01705817      SUPPLIER NUMBER: 03730472     (USE FORMAT 7 OR 9 FOR FULL TEXT)
NAB '85. (convention preview, complete agenda & guide to exhibits)
WORD COUNT:    22650     LINE COUNT:  02819


 34/6/3     (Item 1 from file: 148)
08124425      SUPPLIER NUMBER: 17389671     (USE FORMAT 7 OR 9 FOR FULL TEXT)
Plastics technology: manufacturing handbook & buyers' guide 1995/96.(Buyers
  Guide)
WORD COUNT:   174436     LINE COUNT:  15187


 34/6/4     (Item 2 from file: 148)
02829377      SUPPLIER NUMBER: 04182403     (USE FORMAT 7 OR 9 FOR FULL TEXT)
Should you be your own Bell? (T-1 MUX-Switches)
```

WORD COUNT: 4067 LINE COUNT: 00318
??t34/3,k/all
>>>KWIC option is not available in file(s): 621

34/3,K/1 (Item 1 from file: 16)
DIALOG(R)File 16:IAC PROMT(R)

05960064
TTC announces new GSM Option for the T-BERD 107A.

Business Wire Jan 29, 1996 p. 01291420
FULL TEXT AVAILABLE IN FORMAT 7 OR 9 WORD COUNT: 327

...BERD 107A.
This handheld T-Carrier analyzer can now decode and monitor the quality of compressed voice signals in GSM wireless networks. This product will be shown for the first time at the ComNet '96 trade show in Washington, D.C.
The new option allows users...

... the new option enables users to insert a GSM specific VF tone into a single subchannel of a DSO.
"TTC realizes that wireless service providers have special needs for their wireline...

...at any location along the network o Select the DSO and any of the four subchannels they wish to decode and monitor
o Insert a GSM-specific VF tone into single subchannel of a DSO o View activity on a selected channel even when no speech is...

34/3,K/4 (Item 2 from file: 148)
DIALOG(R)File 148:IAC Trade & Industry Database

02829377 SUPPLIER NUMBER: 04182403 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Should you be your own Bell? (T-1 MUX-Switches)
Stix, Gary
Computer Decisions, v18, p74(7)
March 25, 1986
ISSN: 0010-4558 LANGUAGE: ENGLISH RECORD TYPE: FULLTEXT
WORD COUNT: 4067 LINE COUNT: 00318

... bandwidth of 1.54 Mbps. They also allow for a greater volume of voice traffic. Compressed voice signals can be carried in as little as a quarter of the 64-Kbps bandwidth normally...

...stronger bargaining position with competitive vendors of T-1 lines. He says that at the time of the AT&T divestiture two years ago, the bank was quoted an 18-month...credibility challenge, the longer the wait, the more the likelihood that the manager --perhaps, this time , in disguise-- will have to go back to the brass yet again to authorize a... random burst errors occur on T-1 lines. "They occur in some relation to the time of year, in relation to humidity in certain parts of the Southwest, and when a...
...When an error burst occurs, a T-1 max/switch can shut itself off, or " time out,' after an interval. Dave Stewart, GE's manager of programs in

corporate telecommunications operations...

...an option called Customer-Controlled Reconfiguration (CCR) that enables the customer to switch 64-Kbps subchannels in the T-1 link. The switching takes place at an AT&T central office...
?

```
Set     Items   Description
S1     793853   VOICE OR AUDIO OR ANALOG?
S2      26348   S1(2N)SIGNAL?
S3    2261134   COMPRESS? OR REDUC? OR REDN?
S4       1922   SUBCHANNEL? OR SUBCIRCUIT? OR SUBFACILIT? OR SUBLINE? OR S-
              UBPATH? OR SUBLINK? OR SUB()(CHANNEL? OR CIRCUIT? OR FACILIT?
              OR LINE? OR PATH? OR LINK?)
S5    3842877   TIME

S12        25   (BANDWID? OR BAND(1N)WIDTH)(4N)S3 AND S4
S13         0   (VOCOD? OR COMPANDOR?)(4N)S3 AND S4

S36        17   (BANDWID? OR BAND(1N)WIDTH?)(4N)S3(50N)S4
S37         9   S36(50N)(S1 OR S5)
S38         8   RD S37 (unique items)
?
t38/3,k/all
>>>KWIC option is not available in file(s): 621
```

38/3,K/1 (Item 1 from file: 148)
DIALOG(R)File 148:IAC Trade & Industry Database

03652500 SUPPLIER NUMBER: 06977005 (USE FORMAT 7 OR 9 FOR FULL TEXT)
T1 lines offer flexibility, affordability and efficiency. (last in a series)
Kraska, Paul A.
Government Computer News, v7, n19, p39(2)
Sept 12, 1988
ISSN: 0738-4300 LANGUAGE: ENGLISH RECORD TYPE: FULLTEXT; ABSTRACT
WORD COUNT: 1630 LINE COUNT: 00136

... error conditions and running diagnostics.
Five features point-to-point multiplexers have in common are voice compression , subchannel multiplexing to make efficient use of bandwidth for data communications, D4 and DACS compatibility, channel diagnostics and a software-controlled architecture.
Products...

38/3,K/2 (Item 1 from file: 275)
DIALOG(R)File 275:IAC(SM) Computer Database(TM)

01529543 SUPPLIER NUMBER: 12487925 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Routing: What's around the bend? (Cover Story)
Petrosky, Mary
LAN Technology, v8, n8, p50(11)
August, 1992
DOCUMENT TYPE: Cover Story ISSN: 1042-4695 LANGUAGE: ENGLISH
RECORD TYPE: FULLTEXT; ABSTRACT
WORD COUNT: 4225 LINE COUNT: 00329

... Network managers can use a T1 line more efficiently by connecting a router to a time -division multiplexer. A TDM can handle both analog and digital traffic, and provides bandwidth management through techniques such as multiplexing, data compression , and dividing a link into subchannels .
"Router technology is going to be combined in the traditional TDM for

network management and...

38/3,K/3 (Item 2 from file: 275)
DIALOG(R)File 275:IAC(SM) Computer Database(TM)

01259991 SUPPLIER NUMBER: 07210939 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Security Pacific relates technology to people. (Security Pacific Bank Corp.) (includes related article on merging acquired banks with Security Pacific) (company profile)
Van Collie, Shimon-Craig
Computers in Banking, v5, n12, p24(5)
Dec, 1988
DOCUMENT TYPE: company profile ISSN: 0742-6496 LANGUAGE: ENGLISH
RECORD TYPE: FULLTEXT; ABSTRACT
WORD COUNT: 2601 LINE COUNT: 00206

... any circumstances.
Dorf adds that the direction this technology is heading includes using the network subchannels to carry voice and data. The goal is to subdivide the band widths and digitize voice , pictures, and data. As a result, other cost saving features become available such as video conferencing, which reduces traveling expenses. Another strong area of development is high-speed image technology. Management estimates they...

38/3,K/4 (Item 1 from file: 624)
DIALOG(R)File 624:McGraw-Hill Publications

0704315
THE NEW PC: Four new technologies will usher your next PC into the '90s
BYTE October, 1995; Pg 52; Vol. 20, No. 10
Journal Code: BYTE ISSN: 0360-5280
Section Heading: COVER STORY
Word Count: 2,523 *Full text available in Formats 5, 7 and 9*

BYLINE:
TOM R. HALFHILL

TEXT:
...on the type of universal asynchronous receiver/transmitter (UART) in the computer. USB's maximum bandwidth is 12 Mbps--although actual data throughput is more like 8 Mbps--including a 1-Mbps subchannel for low-speed devices like a mouse or keyboard. That's enough bandwidth to handle everything from keyboards and mice to video monitors, modems, scanners, printers, ISDN adapters, and MPEG-2 compressed video.

In addition, USB is both asynchronous and isochronous. Isochronous transfers, such as audio and video, get top priority, assuring that time -sensitive data streams are not interrupted. USB lets you daisy-chain up to 127 devices...

38/3,K/5 (Item 2 from file: 624)
DIALOG(R)File 624:McGraw-Hill Publications

0088907
T1 network design and -planning made easier: A few simple steps, with just a little math, can greatly simplify the design of a reliable and cost-efficient T1 network.
Data Communications September, 1988; Pg 199; Vol. 17, No. 10
Journal Code: DC ISSN: 0363-6399
Section Heading: Features
Word Count: 2,483 *Full text available in Formats 5, 7 and 9*

BYLINE:
Ranjana Sharma, Telesyne Consulting Inc., San Jose, Calif.

TEXT:
... which can be grouped or further submultiplexed to transport the following special services:

Forty-four compressed 32-kbit/s ADPCM (adaptive differential pulse code modulated) voice channels (four bundles of six contiguous DS-0s, each bundle carrying 11 voice channels plus one signaling channel).

Subrate data channels, consisting of multiple data channels of four...
...4, 4.8, 9.6, and 56 kbit/s--which fit into one DS-0 subchannel .

Compressed video, which typically occupies one bundle (384 kbit/s), equivalent to six PCM or 11 ADPCM channels.

A physical T1 backbone supports a logical network of lower bandwidth links. This mapping of the logical network onto the physical network actually determines the number...

38/3,K/6 (Item 3 from file: 624)
DIALOG(R)File 624:McGraw-Hill Publications

0069403
Pondering the private-to-public fiber connection: FDDI to Sonet
Data Communications May, 1988; Pg 68; Vol. 17, No. 5
Journal Code: DC ISSN: 0363-6399
Section Heading: Newsfront
Word Count: 597 *Full text available in Formats 5, 7 and 9*

BYLINE:
Robert Rosenberg

TEXT:
... the result of tokens taking too long to circulate over long distances, is likely to reduce the throughput when intercity FDDI rings are created.

Another drawback of the bridge approach is that much of the Sonet subchannel bandwidth dedicated to FDDI would go unused most of the time , since FDDI traffic tends to remain in the local area.

FDDI and Sonet could also...

38/3,K/7 (Item 4 from file: 624)
DIALOG(R)File 624:McGraw-Hill Publications

0003843
Long overdue, T1 takes off --but where is it heading?: After 20 years, T1 is moving quickly into customers' premises. Indeed, even ISDN is paying attention.
Data Communications June 1985; Pg 120; Vol. 14, No. 7
Journal Code: DC ISSN: 0363-6399
Section Heading: Feature
Word Count: 10,893 *Full text available in Formats 5, 7 and 9*

BYLINE:
Edwin E. Mier, DATA COMMUNICATIONS

TEXT:
...multiplexing, called 4 multiplexing, effective this past January 5. With 4, up to 44 customer voice conversations can be carried over the same DS-1 signal that earlier could support only 24.

4 digitizes a voice signal using 32-kbit/s ADPCM (adaptive differential pulse code modulation), which cuts in half the 64-kbit/s bandwidth otherwise required for voice digitization. ADPCM is not simply a halving of the 64-kbit/s rate, however, but rather a type of voice -coding compression . As a result, 4 multiplexing limits data transmission over any of the subchannels to no more than about 4.8 kbit/s, or about what a user would get from a voice -grade line. By comparison, users can transmit up to 56 kbit/s in a T1...

38/3,K/8 (Item 1 from file: 636)
DIALOG(R)File 636:IAC Newsletter DB(TM)

01523219
CABLELABS RFP ENTERS CONTRACTING PHASE; PROMOTES DIGITAL VIDEO TECHNOLOGY
Video Technology News June 22, 1992 V. 5 NO. 13
ISSN: 1040-2772 WORD COUNT: 583
PUBLISHER: Phillips Publishing, Inc.

... U.S. HDTV standard, the system includes satellite integrated receiver/decoders (IRDs) that can receive analog as well as digital transmissions. Allows up to 10:1 compression of NTSC signals. The proposal will include cable consumer decompression units. (312/541-5030.)

Philips Broadband Networks--Its compressed Digital Video System includes an "open" architecture that will allow multiple vendors. The system uses MPEG video compression , the European Musicam audio compression standard and the Eurocrypt conditional access system with smartcard technology. It allows up to 4,096 digital subchannels within each 6 MHz of bandwidth and is designed to accommodate NTSC as well as 16:9 HDTV signals. (315/682-9105.)

Scientific-Atlanta/Zenith Electronics Corp.--The system combines S-A's vector quantization (VQ) compression technology with the 4-VSB modulation and transmission technology that is used in Zenith's...
?

Set	Items	Description
S1	793853	VOICE OR AUDIO OR ANALOG?
S2	26348	S1(2N)SIGNAL?
S3	2261134	COMPRESS? OR REDUC? OR REDN?
S5	3842877	TIME
S6	597	SINGLE(1W)(SIDEBAND? OR SIDE()BAND? ?)
S20	19677	BASESTATION? OR LANDSTATION? OR (BASE OR LAND)()STATION?
S21	196	S20 AND S1(4N)S3 AND S5
S22	0	S20 AND S1(4N)S3 AND S6

```
Set     Items    Description
S1      793853   VOICE OR AUDIO OR ANALOG?
S2       26348   S1(2N)SIGNAL?
S3     2261134   COMPRESS? OR REDUC? OR REDN?

S6         597   SINGLE(1W)(SIDEBAND? OR SIDE()BAND? ?)

S39       4352   S1(4N)S3(50N)RECEIV?
S40          8   S39(50N)S6
S41          7   RD S40 (unique items)
?
t41/3,k/all
>>>KWIC option is not available in file(s): 621
```

41/3,K/1 (Item 1 from file: 15)
DIALOG(R)File 15:ABI/INFORM(R)

00927857 95-77249
Put on the squeeze
Shapiro, Norman
Communications v31n9 PP: 48 Sep 1994
ISSN: 0010-356X JRNL CODE: CMN
AVAILABILITY: Fulltext online. Photocopy available from ABI/INFORM 8074.00
WORD COUNT: 1033

...TEXT: sideband, what kind of modulation is it? Actually, the modulation method is called amplitude compandored single sideband (ACSB) with reduced carrier. The transmit modulation is compressed to maximize the average RF power out. The more power, the better the signal-to-noise ratio. At the receive end, the recovered audio is decompressed back to its natural sound.

The microphone's output is compressed and passed to an analog -to-digital converter. The converter's output is PCM (pulse coded modulation) and it is...

... in the DSP with the signal in this digital form. The microphone signal is further compressed , pre-emphasis is added, and the higher frequency components of the microphone's signal are...

41/3,K/2 (Item 1 from file: 148)
DIALOG(R)File 148:IAC Trade & Industry Database

04822956 SUPPLIER NUMBER: 08888288 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Novatel looking to enter land mobile radio.
Industrial Communications, n35, p5(2)
Sept 7, 1990
ISSN: 0737-0415 LANGUAGE: ENGLISH RECORD TYPE: FULLTEXT
WORD COUNT: 983 LINE COUNT: 00075

... s Gary H. Bolen extolled the virtues of narrowband technology for Public Safety.

A small nucleus of people is currently involved in the planning process at NovAtel, which is focusing on...

...several taxi and trucking companies. Texas Utilities, Commonwealth Edison, New York Power and Alabama Power are a few of the utilities involved. But only 2% of the current narrowband users are...

...He separated narrowband into 2 different categories, amplitude compandored single sideband (ACSSB) and real zero single sideband (RZSSB). "Narrowband technologies are focused strict on single channel, 5 kHz wide separation," he said. "Typically, the technology is based on amplitude-modulated single sideband techniques.

"Narrowband technology eliminates the carrier and the lower sideband, broadcasting only on the upper sideband. This sideband contains the audio information, and the result is a 5 kHz channel separation," he added.

Most, if not all, of the narrowband technology in use today is ACSSB. ACSSB is a combination of compression and expansion of the signal. The transmitter can compress the signal, if needed, and the receiver expands the signal to its original size on the other side. "The result is you...

41/3,K/3 (Item 2 from file: 148)
DIALOG(R)File 148:IAC Trade & Industry Database

04112295 SUPPLIER NUMBER: 07976819 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Radio data transmission techniques for SCADA; how Enron uses meteor burst radio for reliable master station communications with remote radio points.
Frasier, William E.
Pipe Line Industry, v71, n3, p40(3)
Sept, 1989
ISSN: 0032-0145 LANGUAGE: ENGLISH RECORD TYPE: FULLTEXT
WORD COUNT: 1533 LINE COUNT: 00118

... in carrier or mirror image sideband signals. It is very much based upon late 1940 single sideband techniques with some modern refinement. First the input data or voice is compressed as was done in the broadcast industry to maintain nearly full modulation without overloading. Then a pilot tone is added for the receiver to lock-on in order to detect the signal properly.

There was some ACSB activity

41/3,K/4 (Item 3 from file: 148)
DIALOG(R)File 148:IAC Trade & Industry Database

04070416 SUPPLIER NUMBER: 07783865 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Spectrum allocation forthcoming ... any for cellular?
Industrial Communications, n26, p6(3)
July 21, 1989
ISSN: 0737-0415 LANGUAGE: ENGLISH RECORD TYPE: FULLTEXT
WORD COUNT: 887 LINE COUNT: 00073

... is much easier to understand. The "amplitude" part of the term indicates the technique is analog , not digital.

"The term 'compand' is the combination of 2 other words: COMpress and exPAND...Companding is achieved by first compressing the amplitude of the voice signal before modulation and transmission, and then expanding this signal after detection in the receiver to restore its original dynamic range," according to the Justice report.

'Sideband' is a generic...

...sideband amplitude modulation with suppressed carrier (SSBSC) has the narrowest emission bandwidth of all the analog modulation schemes. Companding is an enhancement to SSBSC to make it even more efficient.
"At...

41/3,K/5 (Item 4 from file: 148)
DIALOG(R)File 148:IAC Trade & Industry Database

02173401 SUPPLIER NUMBER: 03477024 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Spectrum-based communications. (part 3 of a series on developments in telecommunications)
Stewart, Alan
Telephone Engineer & Management, v88, p76(6)
Oct 15, 1984
ISSN: 0040-263X LANGUAGE: ENGLISH RECORD TYPE: FULLTEXT
WORD COUNT: 3045 LINE COUNT: 00249

... has exceeded projections to the extent that the Federal Communications Commission (FCC) has adopted a reduced orbital criteria to avoid impending congestion. This new criteria has major implications both for control...

...a system of time-sharing the full transponder bandwidth among group of earth stations enables single sideband modulation to be used effectively.
Another major step toward full implementation of bandwidth-limited modulation...

41/3,K/6 (Item 1 from file: 624)
DIALOG(R)File 624:McGraw-Hill Publications

0110678
Network impairments: The dark side of the force
Data Communications February, 1989; Pg 122; Vol. 18, No. 2
Journal Code: DC ISSN: 0363-6399
Section Heading: Features
Word Count: 1,091 *Full text available in Formats 5, 7 and 9*

TEXT:
...to have a periodic, time-varying amplitude level.

Cause: Residual amplitude modulation after demodulation of single -sideband amplitude modulation in carrier systems. N-carrier systems are the most common source.

Effect...

..., adapt to and cancel this impairment. Although generally not very prevalent, amplitude jitter can significantly reduce receiver error margin.

Echo: Also referred to as reflection, this impairment occurs at multiple points in the network and causes the modem receiver to see multiple, attenuated, delayed, and distorted versions of its own transmitted and received signal.

Cause: Echo is caused by any analog impedance mismatch in the network. The largest echo with which a modem must deal is...

41/3,K/7 (Item 1 from file: 636)
DIALOG(R)File 636:IAC Newsletter DB(TM)

00666079
DESPERATELY SEEKING SPECTRUM
INDUSTRIAL COMMUNICATIONS July 21, 1989 V. 1 NO. 28
ISSN: 0737-0413 WORD COUNT: 549
PUBLISHER: Phillips Publishing, Inc.

...is much easier to understand. The "amplitude" part of the term indicates the technique is analog , not digital.

"The term 'compand' is the combination of 2 other words: COMpress and exPAND...Companding is achieved by first compressing the amplitude of the voice signal before modulation and transmission, and then expanding this signal after detection in the receiver to restore its original dynamic range," according to the Justice report.

Sideband' is a generic...

... sideband amplitude modulation with suppressed carrier (SSBSC) has the narrowest emission bandwidth of all the analog modulation schemes. Companding is an enhancement to SSBSC to make it even more efficient.
"At...
?

```
S4       1922   SUBCHANNEL? OR SUBCIRCUIT? OR SUBFACILIT? OR SUBLINE? OR S-
                UBPATH? OR SUBLINK? OR SUB()(CHANNEL? OR CIRCUIT? OR FACILIT?
                OR LINE? OR PATH? OR LINK?)
S5    3842877   TIME

S42    927116   PAGE OR RADIOPAG? OR PAGING OR BEEP? OR PAGE? ? OR RADIO? -
                ?(2N)(SUMMON? OR PRECONI?)
S43        59   S42(50N)S4
S44         8   S43(50N)(S5 OR AMPLI? OR FILTER?)
S45         5   RD S44 (unique items)
?
t45/3,k/all
>>>KWIC option is not available in file(s): 621
```

45/3,K/1 (Item 1 from file: 15)
DIALOG(R)File 15:ABI/INFORM(R)

00736886 93-86107
IBM woos large PC sites with extensive service
Corcoran, Cate
InfoWorld v15n27 PP: 1, 79 Jul 5, 1993
ISSN: 0199-6649 JRNL CODE: IFW
AVAILABILITY: Fulltext online. Photocopy available from ABI/INFORM 12701.01
Article Ref. No.: B-IFW-290-1
WORD COUNT: 390

...TEXT: he said.

The company also announced it will break its PS/2 line into two sublines : Traditional and Enhanced.

The Traditional line is aimed at the same corporate sites as the...

... 2 Enhanced line is aimed at "early adopters" who want the newest technology. (See chart, page 1.)(Chart omitted)

With its Premium Partners Program, IBM intends to address the unique needs and problems IS managers face when companies install thousands of PCs at one time for specific mission-critical applications.

Lately, IBM has relied on its dealers to meet the...

45/3,K/2 (Item 1 from file: 148)
DIALOG(R)File 148:IAC Trade & Industry Database

07877787 SUPPLIER NUMBER: 16882249 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Universal bus to simplify PC I/O: new interface to service keyboards, mice, telecom, and more. (12-Mbps Universal Serial Bus backed by Intel, Microsoft and computer and telephony makers)(includes related article on how to obtain the specification)
Slater, Michael
Microprocessor Report, v9, n5, p1(5)
April 17, 1995
ISSN: 0899-9341 LANGUAGE: ENGLISH RECORD TYPE: FULLTEXT
WORD COUNT: 3995 LINE COUNT: 00307

... version of the specification has just been released, and it is available on request (see page 9). After collecting industry feedback, the USB group plans to deliver version 1.0 in...

....debut at fall Comdex and begin shipping in the first quarter of 1996.
Low-Speed Subchannel Cuts Peripheral Cost
Early USB design work assumed a data rate of 5 Mbps, representing...

...signal quality is significant for a low-cost device like a mouse. At the same time , some high-end applications--notably MPEG-2 video and other CD-ROM software would benefit...

45/3,K/3 (Item 1 from file: 275)
DIALOG(R)File 275:IAC(SM) Computer Database(TM)

01686904 SUPPLIER NUMBER: 16000104 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Star-dot-star tips. (question-and-answer)(Here's How) (Column)
Glass, Brett
PC World, v12, n6, p282(3)
June, 1994
DOCUMENT TYPE: Column ISSN: 0737-8939 LANGUAGE: ENGLISH
RECORD TYPE: FULLTEXT
WORD COUNT: 1823 LINE COUNT: 00138

... Timer from the Object list and Timer() from the Proc list, and enter If Mid$(Time $,4,4) = "00:0" Then Beep between the Sub and End Sub lines . This single line does all the real work in the program: Once every second, it checks four characters in the middle of the Time $ function's output (which is formatted HH:MM:SS) to see if the time is within the first 10 seconds after the hour. If it is, the speaker beeps...

45/3,K/4 (Item 1 from file: 621)
DIALOG(R)File 621:IAC New Prod.Annou.(R)

0353027
News Release
DATELINE: BROOMFIELD, CO April 5, 1993 WORD COUNT: 1106

McDATA ANNOUNCES UPGRADES FOR THE MARKET'S FIRST PORTABLE ESCON PROTOCOL ANALYZER

45/3,K/5 (Item 1 from file: 636)
DIALOG(R)File 636:IAC Newsletter DB(TM)

02329636
BROOKTROUT NETWORKS FORMS: IS THERE A WAN FAX MARKET?
Electronic Mail & Messaging Systems September 1, 1993 V. 17 NO. 17
ISSN: 8756-2537 WORD COUNT: 1723
PUBLISHER: BRP Publications

... justification due to higher phone rates and a high incidence of fax usage due to time -zone differences. Currently, X.25 and frame relay networks seldom are used for fax transmissions.

Interestingly, many fax pages are currently sent by default over T1 networks as voice traffic. But BNGI sees this as a viable market, largely because most fax travels over T1 _subchannels_ very inefficiently - often at only 4.8 kbps and at best at 9.6 kbps...

...travel six to 12 times faster using DAFS, or six to 12 times as many _pages_ could use the same amount of bandwidth. Efficiency is the T1 rub, because fax pages...
?

```
S30    324429   DEMODULAT? OR DE()MODULAT? OR MODEM? OR S1(3N)DIGIT?
S46     53680   S30(50N)(TERMINAL? OR RECEIV?)
S47       537   S46(5N)FILTER?
S48       143   S47(50N)(MODULAT? OR QAM)
S49        51   S48(50N)AMPLI?
S50        26   S49(50N)SIGNAL?(2N)PROCESS?
S51        19   RD S50 (unique items)
S52         2   S51/1996
S53        17   S51 NOT S52
?
t53/3,k/all
>>>KWIC option is not available in file(s): 621
```

53/3,K/1 (Item 1 from file: 15)
DIALOG(R)File 15:ABI/INFORM(R)

00969981 96-19374
Investigating the cutting edge of radio technology
Kelly, Kevin
Telesis n99 PP: 21-30 Dec 1994
ISSN: 0040-2710 JRNL CODE: TLS
AVAILABILITY: Fulltext online. Photocopy available from ABI/INFORM 16018.00
WORD COUNT: 5621

...TEXT: for evaluation. And because the system would have to operate under a complex set of signaling conditions, the problem could not be solved by purely analytic solutions.

The ATC's solution...

... radio link elements and any inherent component imperfections. The simulation included all critical transmitter and receiver components and processes, such as amplifiers, reference oscillators, modulators, filters, coders, and demodulators. Team members built enough flexibility into the simulation to allow experimentation with different technologies, components...

53/3,K/2 (Item 1 from file: 16)
DIALOG(R)File 16:IAC PROMT(R)

05750991
HARRIS SEMICONDUCTOR INTRODUCES INDUSTRY'S FIRST RF-THROUGH-BASEBAND DATA RADIO SOLUTION FOR 2.4-GHz, WIRELESS LAN CARDS IN PCMCIA TYPE-II FORM FACTOR

PR Newswire Oct 2, 1995 p. 1002FL015
FULL TEXT AVAILABLE IN FORMAT 7 OR 9 WORD COUNT: 1452

... services. These include Utilicom (Goleta, Calif.), L.S. Research (Cedarburg, Wisc.), Sigtec (Columbia, Md.) and Signal Technologies (Longwood, Fla.).

CHIPSET DETAILS

The PRISM chipset comprises four tightly matched ICs: the industry's only DSSS baseband processor with on-board A/D (analog -to-digital) converters and separate DSSS spreader and de-spreader; a highly integrated IF/QMODEM with modulator and demodulator , plus limiter, programmable low-pass filters , and RSSI (received signal strength indication) on one chip, a silicon 2.4 GHz RF/intermediate frequency (IF) converter and a silicon RF power amplifier . These last two ICs typically must be manufactured with GaAs processes to archive necessary frequency performance. To fabricate them in silicon for 2.4 - 2.5 ...

53/3,K/3 (Item 2 from file: 16)
DIALOG(R)File 16:IAC PROMT(R)

05463698
BROADCOM HAS QUADRATURE AMPLITUDE MODULATION RECEIVER OFFERING 40MBPS TRANSMISSION SPEED

Computergram International December 16, 1994 p. N/A
ISSN: 0268-716X
FULL TEXT AVAILABLE IN FORMAT 7 OR 9 WORD COUNT: 215

Broadcom Corp, Los Angeles has announced the 64/256 Quadrature Amplitude Modulation digital-transmission receiver for cable television applications. The QAMLink architecture, initially developed with Scientific ...

... The BCM3100 integrates these three chips into a low-cost CMOS chip. In 256-Quadrature Amplitude Modulation mode, the BCM3100 QAMLink Receiver achieves digital transmission speeds of 40Mbps. This increases the coaxial cable channel capacity 10 to 25 times. Using proprietary signal processing algorithms, circuit designs and layout techniques, Broadcom integrated Quadrature Amplitude Modulation demodulation , Nyquist filtering , carrier and timing synchronisation and adaptive equalisation into a single low-cost chip. Operating in 64-Quadrature Amplitude Modulation mode, the BCM3100 enables a standard 6MHz analogue channel to carry 30Mbps of digital data. With 64 -Quadrature Amplitude Modulation , cable operators can send around 20 movies using MPEG I compression and 10 movies using...

... the effects of channel distortions often found in cable television systems. An adaptive equaliser requires signal processing power of about 1,000m operations per second to blindly adapt to the channel distortions...

53/3,K/4 (Item 3 from file: 16)
DIALOG(R)File 16:IAC PROMT(R)

04395118
Stop the Press!

Telephone Engineer & Management April 15, 1993 p. 64
ISSN: 0040-263X
FULL TEXT AVAILABLE IN FORMAT 7 OR 9 WORD COUNT: 1729

... associated PCS coverage environments. This product digitizes the entire cellular band to transport the RF signal over fiber. Usually starting at a combiner with the radio base station equipment, a high performance digital tuner captures all desired signals , and a mixer provides very high dynamic range.

The entire cellular band then is digitized in preparation for transport. Significant filtering achieving sharp selectivity occurs prior to analog /digital conversion. The signal is converted to optics, and a digitally modulated laser ensures signal delivery without degradation over an extended distance (up to 65% more fiber range as compared with analog transport). The system also employs wave-division multiplexing technology to transmit and receive on a single fiber. The entire cellular band then is transported digitally over the fiber...
...less than a four-microsecond delay.
Remote transceivers at the serving site then demultiplex the signals and reverse the process , going from optical to electrical and digital to analog . A high performance linear power amplifier provides the required RF output power at the antenna. On the uplink, additional filtering occurs prior to digitization. Alarm and order wire information are inserted on the optical bit stream and received by the host equipment.
Flexible network planning for radio interface links. CityCell also addresses other...

53/3,K/5 (Item 4 from file: 16)
DIALOG(R)File 16:IAC PROMT(R)

02528747
Fax/data modem chip set
Exar: Develops XR2900, -01, -02, fax/data modem chip set

Electronic Engineering Times April 09, 1990 p. 52
ISSN: 0192-1541
FULL TEXT AVAILABLE IN FORMAT 7 OR 9 WORD COUNT: 463

...two-chip set is aimed at personal computer and laptop applications.
While the XR2901 digital signal processor chip supports modulation / demodulation functions, the XR2902 is the analog front end, comprising mixed analog and digital circuits. The analog portion consists of the transmit/ receive filters , analog -to-digital and digital-to-analog functions, transmit level attenuator, and programmable gain amplifer . The digital section encompasses interface circuit for the host controller, async/sync converter, transmit clock, and receive clock digital phase-locked loop. The fax/data modem set implements Exar's 2-micron, double-poly, single-metal CMOS process to achieve high integration at low power. Typical combined power consumption for the set is
...two-chip set is aimed at personal computer and laptop applications.
While the XR2901 digital signal processor chip supports modulation / demodulation functions, the XR2902 is the analog front end, comprising mixed analog and digital circuits. The analog portion consists of the transmit/ receive filters , analog -to-digital and digital-to-analog functions, transmit level attenuator, and programmable gain amplifer . The digital section encompasses interface circuit for the host controller, async/sync converter, transmit clock, and receive clock digital phase-locked loop. The fax/data modem set implements Exar's 2-micron, double-poly, single-metal CMOS process to achieve high integration at low power. Typical combined power consumption for the set is

53/3,K/6 (Item 5 from file: 16)
DIALOG(R)File 16:IAC PROMT(R)

2524574
ata/Fax Modem ICs: Exar's Chip Set Can Send/Receive
Exar: New fax/data modem chip set is aimed at PCs & laptops

lectronic Buyers' News April 02, 1990 p. 19
SSN: 0164-6362
FULL TEXT AVAILABLE IN FORMAT 7 OR 9* WORD COUNT: 318

...times and documentation merge.
The two-chip set consists of the XR-2901, a digital signal processor chip supporting primarily the modulation / demodulation function, and the XR-2902, a combination analog -and- digital chip.
The analog portions of the 2902 support the transmit and receive filters , A/D and D/A functions, transmit-level attenuator nd progammable-gain amplifier . The digital protion of the 2902 upports the transmit-and receive -clock generator, async/sync buffer etween the 2901 and host controller and a receive -clock digital hase-locked loop.
The XR-2091 and XR-2902 are available in 40...

53/3,K/7 (Item 1 from file: 148)
DIALOG(R)File 148:IAC Trade & Industry Database

08436017 SUPPLIER NUMBER: 17873127 (USE FORMAT 7 OR 9 FOR FULL TEXT)
The road to conversion. (wireless communications)
Parry, Simon
Electronics Weekly, n1748, p30(1)
Dec 6, 1995
ISSN: 0013-5224 LANGUAGE: English RECORD TYPE: Fulltext; Abstract
WORD COUNT: 1379 LINE COUNT: 00116

... drawbacks - of the heterodyne approach.
The single-chip IC needs the addition of a power amplifier , LNA and filters to implement a GSM terminal 's radio section. The chip integrates two fixed-frequency synthesisers (for the transmit and receive IFs), a programmable frequency-agile UHF synthesiser (for channel selection), RF mixer, single IF down conversion with quadrature demodulation and variable gain and a direct up modulator .
In a superheterodyne receiver the RF spectrum is translated or mixed to an IF before the demodulation is performed. In an FM superhet receiver this latter stage is accomplished by a frequency discriminator at the IF.
The advantage in shifting the modulated spectrum to an intermediate (usually lower) frequency is that the receiver 's selectivity can easily be defined by a fixed-frequency bandpass filter at the IF and by varying the local oscillator's frequency.
A secondary advantage is that, because the demodulation is usually performed at a lower frequency, the IF amplifiers and frequency discriminator are easier to implement.
The drawback of the superhet receiver is that the mixing process is inherently susceptible to any interfering signal at the image frequency (F.sub.C) (+ or -) 2(F.sub.LO). This image is...

53/3,K/8 (Item 2 from file: 148)
DIALOG(R)File 148:IAC Trade & Industry Database

07714287 SUPPLIER NUMBER: 16683510 (USE FORMAT 7 OR 9 FOR FULL TEXT)
The future of RF and analogue. (radio frequency)
Electronics Weekly, n1711, p16(1)
Feb 15, 1995
ISSN: 0013-5224 LANGUAGE: ENGLISH RECORD TYPE: FULLTEXT; ABSTRACT
WORD COUNT: 635 LINE COUNT: 00052

... considered, perhaps brave, examination of the benefits and pitfalls of the technique.

In a superheterodyne receiver , for instance, the RF spectrum is translated or mixed to an intermediate frequency (IF) before the demodulation is performed. In an FM superhet receiver this latter stage is accomplished by a frequency discriminator at the IF. The advantage in shifting the modulated spectrum to an intermediate (usually lower) frequency is that the receiver 's selectivity can easily be defined by a fixed frequency bandpass filter at the IF and by varying the local oscillator's frequency. A secondary advantage is that because the demodulation is usually performed at a lower frequency the IF amplifiers and frequency discriminator are easier to implement.

The drawback of the superhet receiver is that the mixing process is inherently susceptible to any interfering signal at the image frequency (FC[+ or -]2FLO). This image is usually removed by employing an...

...and by having a bandpass (image) filter at the mixer's RF input which rejects signals at the image frequency.

By comparison, in the homodyne, or direct conversion receiver, the information...

53/3,K/9 (Item 3 from file: 148)
DIALOG(R)File 148:IAC Trade & Industry Database

07593892 SUPPLIER NUMBER: 16454654 (USE FORMAT 7 OR 9 FOR FULL TEXT)
IC opens 500-channel frontier to cable systems. (includes related article)
Goldberg, Lee
Electronic Design, v42, n23, p71(6)
Nov 7, 1994
ISSN: 0013-4872 LANGUAGE: ENGLISH RECORD TYPE: FULLTEXT; ABSTRACT
WORD COUNT: 2669 LINE COUNT: 00214

... gain also will amplify any noise present in the channel, significantly lowering the receiver's signal -to-noise ratio.

For this reason, Broadcom developed a circuit called a decision-feed-back...

...coefficients. With the DFE, more rapid and precise compensation can be applied to the incoming signal , allowing for faster response without further amplifying background noise. Under worst-case conditions, a QAM demodulator using this active filtering scheme delivers a bit-error rate (BER) of [10.sup.-5]. By adding Reed-Solomon...

...far better than the [10.sup.-9] required MPEG transmissions.

The BCM3100 features all-digital signal processing . Because the receiver uses an IF-sampled architecture, the IF analog front end is greatly simplified. Quadrature detection is performed digitally, allowing exact I/Q amplitude and phase matching--critically important for QAM systems, in which each part of the quadrature signal contains so much information. Most analog systems require a pilot tone within the channel to help lock the receiver. This wastes precious spectral energy, lowering the effective system signal -to-noise ratio. By performing all acquisition and tracking digitally, Broadcom's receiver can detect...

53/3,K/10 (Item 4 from file: 148)
DIALOG(R)File 148:IAC Trade & Industry Database

06507330 SUPPLIER NUMBER: 14184239 (USE FORMAT 7 OR 9 FOR FULL TEXT)
ISSCC: communications technology. (1993 International Solid State Circuits Conference) (Technical)
Leonard, Milt
Electronic Design, v41, n5, p69(4)
March 4, 1993
DOCUMENT TYPE: Technical ISSN: 0013-4872 LANGUAGE: ENGLISH
RECORD TYPE: FULLTEXT; ABSTRACT
WORD COUNT: 2380 LINE COUNT: 00191

... consists of a direct up-conversion receiver IC, both fabricated with a 12-GHz bipolar process . The 900-MHz quadrature up-conversion modulator performs phase correction. The two-IF receiver includes FM and direct QPSK (DQPSK) demodulators, a 60-dB-dynamic-range AGC amplifier , and an RF mixer with 26-dB attenuation control. The two chips dissipate a total...

...draws 10 [mu]A.
Researchers at the same AT&T Bell Labs facility developed a modem /codec chip for dual-mode cellular telephony. Containing both transmit and receive sections, the chip acts like a modem in the digital mode. The device performs [pi]/4-shifted DQPSK modulation , square-root raised-cosine filtering , and wave-form synthesis. In the analog mode, the chip acts like a codec by performing frequency modulation and analog -to- digital interfacing. The 0.9-[mu]m CMOS device dissipates 100 mW in the digital mode...

...emerging wireless communications systems, the use of image-reject mixers reduces the cost of a receiver by eliminating the need for expensive image filters. The first integrated biCMOS image-reject front...

...Calif. The 2.5-GHz device consists of an image-reject mixer, a low-noise amplifier , bias circuitry, phase-shift elements, and power-down stages. At 1.98 GHz and with a 111-MHz IF signal , image rejection is 14.1 dB, conversion gain is 7.6 dB, and the noise figure is 13 dB. The device is fabricated in a 0.8-[mu]m biCMOS process and consumes 60 mW.
The demand for clearer voice communications and longer transmission ranges for...

53/3,K/11 (Item 5 from file: 148)
DIALOG(R)File 148:IAC Trade & Industry Database

04475295 SUPPLIER NUMBER: 08356032
Fax/data modem chip set. (Exar Corp introduces the FaxPlus XR-2900 chip set) (product announcement)
Bindra, Ashok
Electronic Engineering Times, n585, p52(1)
April 9, 1990

DOCUMENT TYPE: product announcement ISSN: 0192-1541 LANGUAGE: ENGLISH RECORD TYPE: ABSTRACT

...ABSTRACT: V.42/V.42 bis standards. The chips enable the provision of full 9600bps send/ receive fax and 2400bps data (modem) transmissions with the addition of a minimum number of support chips. The XR2901 digital signal processor handles modulation / demodulation functions. The XR2902 is a mixed analog/digital device, whose analog section includes transmit/ receive filters , both digital -to- analog and analog -to- digital conversion, programmable gain amplifier and transmit level attenuator. The digital components include host controller interface, transmit and receive clocks and async/sync converter. Typical power consumption is below 400mW. The XR-2900 is...

53/3,K/12 (Item 6 from file: 148)
DIALOG(R)File 148:IAC Trade & Industry Database

04163302 SUPPLIER NUMBER: 08531851 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Test solutions for GSM. (Groupe Special Mobile)
Whitacre, Jan; Kann, Roger
Communications International, v16, n12, p43(3)
Dec, 1989
ISSN: 0305-2109 LANGUAGE: ENGLISH RECORD TYPE: FULLTEXT
WORD COUNT: 1847 LINE COUNT: 00146

... a measurement technique (known as the Global Test Method) to measure the accuracy of the modulation of the transmitted signal . Different block diagrams to achieve this method can be used.

An analogue approach uses an I and Q (In-phase and Quadrature) demodulator with filtering to produce the necessary signals for measurement of the phase accuracy of the modulated waveform. However, this analogue processing before measurement adds uncertainty to the tests from possible imperfect I/Q balance and filter errors. A digital technique for determining the I/Q components of the received signal would eliminate these errors, offering greater accuracy and the added benefit of a simpler hardware...

...diagram that Hewlett-Packard chose for digitally measuring the phase and frequency errors and the amplitude profile of a continuous-phase modulated RF signal . The modulated RF signal is coupled to a down conversion mixer to provide an intermediate frequency (IF) signal . The IF signal is filtered in an analogue anti-aliasing filter and then digitised to convert the analogue IF signal to a discrete-time data sequence. The sampled data is then sent to a computer where digital signal processing techniques are used to analyse the signal.

From the digitally demodulated data sequence, the phase...

53/3,K/13 (Item 7 from file: 148)
DIALOG(R)File 148:IAC Trade & Industry Database

03920558 SUPPLIER NUMBER: 07323122 (USE FORMAT 7 OR 9 FOR FULL TEXT)
High-resolution ADCs simplify system design in DSP applications. (includes related articles on choosing the right sampling ADC, synchronous versus asynchronous conversion and minimizing the effects of digital feedthrough) (analog-to-digital converters, digital signal processing)

(technical)
Sylvan, John; Malone, Bob; Reidy, John; Paul, Errico
EDN, v34, n10, p155(19)
May 11, 1989
DOCUMENT TYPE: technical ISSN: 0012-7515 LANGUAGE: ENGLISH
RECORD TYPE: FULLTEXT; ABSTRACT
WORD COUNT: 6495 LINE COUNT: 00497

... filter and amplify the input of the receive filter with their on-chip programmable gain amplifiers . Use pins DB.sub.2 through DB.sub.7 on IC.sub.5 to set...

...use pins DB.sub.0 through DB.sub.7 to program the gain of the receive filter , IC.sub.4, in 3-dB increments.

Because high-speed modems use quadrature amplitude modulation , synchronized timing is critical. Errors in the conversion timing produce phase errors and create distortion in the frequency domain, which are unacceptable when you're trying to control phase- modulated baseband signals . Use an external transmit/ receive clock that runs asynchronously to the processor to generate precisely timed conversion-start commands for the A/D converter. (You should also...

...commands, connect a single transmit/receive clock to the filters' SYNCIN terminals. The filters' SYNCOUT signals generate the CONVST and LDAC commands for the converters. You can program one of the...

53/3,K/14 (Item 8 from file: 148)
DIALOG(R)File 148:IAC Trade & Industry Database

02164848 SUPPLIER NUMBER: 03335592 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Breakthrough in digital radio. (quadrature amplitude modulation)
Rockwell, David
Telephone Engineer & Management, v88, p51(3)
July 1, 1984
ISSN: 0040-263X LANGUAGE: ENGLISH RECORD TYPE: FULLTEXT
WORD COUNT: 1744 LINE COUNT: 00135

... FCC requirements. The receiver filter is used to form a Nyquist channel for the regeneration process . This also is the filter that determined noise bandwidth at the generator. What's more, the transmitter filter controls signal peaking from the transmitter amplifier . Both filters, working in tandem, prevent spillage (or interference) into adjacent channels. Typically, channel filtering...

...shape, with half the filtering done at the transmitter and the other half at the receiver .

Filtering in this system is done at baseband level for several reasons: 1) it is independent...

...guaranteed, 3) without requiring special techniques, it remains very stable, and 4) it permits the modulation and demodulation to be independently loopback tested. In-service monitoring

This technology is realized in the two...

53/3,K/15 (Item 1 from file: 621)
DIALOG(R)File 621:IAC New Prod.Annou.(R)

0328485
News Release
DATELINE: SAN JOSE, CA · April 30, 1992   WORD COUNT: 784

EXAR Unveils an Integrated Chip Solution for Fax, Data & Voice with Optimum Windows (R) Operation

53/3,K/16   (Item 1 from file: 624)
DIALOG(R)File 624:McGraw-Hill Publications

0222159
On the Radio : Wireless LANs give new meaning to the term "ether" net
BYTE June, 1990; Pg 224A; Vol. 15, No. 6
Journal Code: BYTE   ISSN: 0360-5280
Section Heading: State of the Art: Networking
Word Count: 3,093   *Full text available in Formats 5, 7 and 9*

BYLINE:
Sorin Davidovici

TEXT:
... many active signals and limits mutual interference. The amplitude, phase, or frequency of a carrier signal is modulated (modified) according to the information to be sent. The receiver detects these changes ...

... will not overlap. Thus, even though the input signals share the same frequency band, the modulated signals are separated in frequency. The frequency separation makes it possible to extract the signal of interest from a multitude of signals active at the same time.

A carrier signal is simply a sine-wave signal . Receivers select specific signals by using a filter tuned to the appropriate frequency. All other signals , which are simply interference, are eliminated. Once the signal is snatched out of the ether, the receiver extracts the information contained in it in a process called demodulation . If the signal in question originates at a commercial FM station, it most likely carries music and speech that are amplified and played through the speakers after being demodulated . In data communications, the data is extracted by the Physical layer and passed on to...

... and FM have found applications in data transmission modems. Figure 2 illustrates AM and FM signals modulated by a 11010 data pattern. Note that both systems transform data represented by 0s and...

...FM. In addition to the carrier, it uses a second auxiliary signal in the modulation process . This signal is a user- or network-specific digital code with a much higher transfer...

53/3,K/17   (Item 2 from file: 624)
DIALOG(R)File 624:McGraw-Hill Publications

0013942
Running Ethernet modems over broadband cable: Broadband promises bigger,

better Ethernets that use controllers now installed in many computers.
This tutorial shows how the modems that run these new networks operate.

Data Communications May, 1986; Pg 199; Vol. 15, No. 5
Journal Code: DC ISSN: 0363-6399
Section Heading: Feature
Word Count: 5,189 *Full text available in Formats 5, 7 and 9*

BYLINE:
Menachem E. Abraham, ChipCom Corp., Needham, Mass.

TEXT:
... amplified the same amount without distortion. Lack of linearity might induce cross-modulation, causing the signals in different bands to affect and interfere with each other. Cross-modulation can result in...

...degradation of video signals or bit errors in data channels.

In order to keep these amplifiers in their linear region, the signals on the broadband coaxial trunk are relatively small, ranging... ...100 mv. Also, to minimize loading of the trunk by the stations attached, the broadband modems are connected to the coaxial cable through a tap that further attenuates this signal to a level typically between 0.5 and 5 mv. The receiving section of the broadband transceiver is, therefore, required to do carrier detection and demodulation on small signals in a relatively wide amplitude range and at very high frequencies. The receiver first amplifies the signal and then converts it down to an intermediate frequency (IF), where it filters out signals from the other bands. Then demodulation and carrier detection functions are performed, also at the IF.

Two different techniques can be used for the demodulation process: synchronous detection and delay-line detection. In the synchronous case, a reference IF carrier is recovered and used to determine the data content of the received phase- modulated IF signal. Delay-line detectors use the phase of the IF carrier in the previous...

...and transmission impairments.

Bit-timing information is typically extracted from the demodulated data waveform by processing the low-to-high or high-to-low signal transitions. The recovered clock is used...
?pause
>>> PAUSE started.
?

```
S30     324429  DEMODULAT? OR DE()MODULAT? OR MODEM? OR S1(3N)DIGIT?

S46      53680  S30(50N)(TERMINAL? OR RECEIV?)

S55        941  S46(50N)FILTER?
S56        227  S55(50N)(MODULAT? OR QAM)
S57         90  S56(50N)AMPLI?
S58         39  S57(50N)SIGNAL?(2N)PROCESS?
S59         29  RD S58 (unique items)

S62          3  S59/1996
S63          9  S59 NOT (S62 OR S50)
?a
t63/3,k/all
>>>KWIC option is not available in file(s): 621
```

63/3,K/1   (Item 1 from file: 9)
DIALOG(R)File   9:Business & Industry(R)

01360758
Telefunken Announce Power-Saving RF Chip Set
(Telefunken Semiconductors introduces chip set for 900-MHz RF applications, including complete front-end and RF receiver)
Electronic Buyers News, p 38
December 18, 1995
DOCUMENT TYPE: Journal; News Brief   ISSN: 0164-6362   (United States)
LANGUAGE: English   RECORD TYPE: Fulltext
WORD COUNT: 333

TEXT:
...with a power amplifier, low-noise amplifier, and antenna switch. Built on a gallium arsenide process , it contributes minimal noise of 1.5 dB from the low-noise amplifier for 45 MHz to 50 MHz and 900-MHz cordless phone designs. The U7001B consumes...

...talk times. It requires few external components and saves space in compact designs.

For the receive function in both voice and data applications, the U3500BM integrates a low-noise amplifier , intermediate frequency amplifier , FM demodulator , and receive - signal strength indicator. The U3500BM features low-pass filters on board and a mixer for scrambling, which can reduce the need for external components.

For the transmit function, the U2891B integrates the modulator and up-conversion features on a single chip, with operation up to 3 gigahertz. The...

63/3,K/2   (Item 1 from file: 15)
DIALOG(R)File   15:ABI/INFORM(R)

00636994   92-51934
TV Technology Goes Digital
Gyorki, John R.
Machine Design   v64n18   PP: 61-65   Sep 10, 1992
ISSN: 0024-9114   JRNL CODE: MDS
AVAILABILITY: Fulltext online. Photocopy available from ABI/INFORM 1075.00

Article Ref. No.: B-MDS-113-20
WORD COUNT: 1809

...TEXT: s favor, the consumer is the ultimate winner. The first HDTVs will be relatively expensive, and it may take HDTV several years to become as widespread as conventional TV, partly because...of transmission is called vestigial sideband. At the transmitter, separate video and audio channels are processed and mixed before transmission. At the receiver, the composite signal is amplified and split bank into separate sound and picture frequencies.

The Channel Compatible DigiCipher HDTV systems proposed by Massachusetts Institute of Technology also uses separate video and audio channels. But digital signal processing handles them far differently than NTSC. For one thing video and auto channels are amplified separately, but are encoded and multiplexed before being modulated and transmitted. The CCD-HDTV system broadcasts both high-definition and conventional signals on the same channel with mutual interference. When the receive selects a channel, it demodulates and demultiplexes the signal , then separates it into video and audio channels for decoding and amplification .

Other systems, such as the Zenith Digital Spectrum Compatible HDTV, broadcast HDTV signals on unallocated channels while simulcasting conventional signals on the usual channels. All proposed American systems use digital signal compression, decompression, filtering and distortion correction to squeeze high-resolution signals within the 6-MHz bandwidth.

THE CHIPS BEHIND THE SCREEN

Prototype HDTV receivers today fill...

63/3,K/3 (Item 1 from file: 16)
DIALOG(R)File 16:IAC PROMT(R)

04959992
Handset split solves foundry problems

Electronics Times February 24, 1994 p. 10
ISSN: 0142-3118
FULL TEXT AVAILABLE IN FORMAT 7 OR 9 WORD COUNT: 143

... of a GSM handset into a silicon bipolar transceiver and synthesiser device for high frequency signals and a CMOS device for lower frequency processing .

This arrangement solves the problem of finding a foundry with a BICMOS process providing a high enough bipolar f(t) for low noise mixers and demodulators capable of 1.9GHz operation.

The researchers suggested this problem would prevent single chip GSM analogue front ends for the foreseeable future.

The bipolar front end consisted of two quadrature modulators , a phase shifter and receiver and transmit mixers and amplifiers .

The CMOS analogue baseband device provided a transmission shaper, radio control for transmit and receive, bandpass filters and amplifiers .

The synthesiser function was split between the two devices, with the high frequency part providing...

63/3,K/4   (Item 2 from file: 16)
DIALOG(R)File 16:IAC PROMT(R)

03467497
Mobility moves into digital arena: ... and with the shift a range of devices is being used by designers to improve reception and cut the cost and size of units.

Electronics Times   October 31, 1991   p. 26
FULL TEXT AVAILABLE IN FORMAT 7 OR 9 WORD COUNT: 1633

...chipsets.
The move to digital techniques makes use of the greater noise immunity of digital signalling , as well as the higher spectrum utilisation allowed by digital protocols and the strength of...

...processing and high density digital circuitry.
All the imminent digital communications protocols use transmit and receive channels. On the subscriber side, a signal goes from microphone or tone dialler through an analogue conditioning section to a digital section, where voice coding and channel coding are performed.
The signal stream is modulated and enters the intermediate and high frequency sections. Finally the signal is amplified and applied to the antenna.
The receive channel operates in the inverse manner, passing through analogue rf, digital and analogue baseband sections before driving the speaker.
The differences between CT2, gsm, Dect and pcn lie in the rf carrier frequency, and the stipulated digital voice and channel coding standards.
Designers have several options for implementation in the signal flow. They include a/d and d/a converters and discrete solutions for amplification and filtering closest to the user, or mixed signal asics. For the dsp section they can use mixed standard or semicustom products.
Outside the signal channels, there is the need for control and interfacing to a keypad and to lcd
...chipsets.
The move to digital techniques makes use of the greater noise immunity of digital signalling , as well as the higher spectrum utilisation allowed by digital protocols and the strength of...

...processing and high density digital circuitry.
All the imminent digital communications protocols use transmit and receive channels. On the subscriber side, a signal goes from microphone or tone dialler through an analogue conditioning section to a digital section, where voice coding and channel coding are performed.
The signal stream is modulated and enters the intermediate and high frequency sections. Finally the signal is amplified and applied to the antenna.
The receive channel operates in the inverse manner, passing through analogue rf, digital and analogue baseband sections before driving the speaker.
The differences between CT2, gsm, Dect and pcn lie in the rf carrier frequency, and the stipulated digital voice and channel coding standards.
Designers have several options for implementation in the signal

flow. They include a/d and d/a converters and discrete solutions for amplification and filtering closest to the user, or mixed signal asics. For the dsp section they can use mixed standard or semicustom products.
Outside the signal channels, there is the need for control and interfacing to a keypad and to lcd...

63/3,K/5 (Item 1 from file: 148)
DIALOG(R)File 148:IAC Trade & Industry Database

07593891 SUPPLIER NUMBER: 16454656 (USE FORMAT 7 OR 9 FOR FULL TEXT)
A closer look at QAM (quadrature-amplitude modulation)
Samueli, Henry
Electronic Design, v42, n23, p73(2)
Nov 7, 1994
ISSN: 0013-4872 LANGUAGE: ENGLISH RECORD TYPE: FULLTEXT; ABSTRACT
WORD COUNT: 891 LINE COUNT: 00070

... modem implementations, a good portion of the filtering and equalization functions is performed using digital- signal -processing (DSP) techniques. Therefore, the analog-to-digital converters and the DACs often are located at some IF rather than at baseband.
The low-pass filters in the modulator perform the important function of bandlimiting the transmitted RF spectrum to avoid any adjacent-channel interference. However, the process of filtering the square-topped pulses at the outputs of the DACs causes these pulses to smear and potentially interfere with one another. Fortunately, there exists a special class of low-pass filters , known as Nyquist filters , which can generate a filtered pulse without the normal smearing. Consequently, all high-performance QAM modems use Nyquist filters in their transmitter and receivers .
Each symbol contains M bits of information. M/2 bits are used to amplitude - modulate the cosine carrier and M/2 bits are used to amplitude - modulate the sine carrier. Because R symbols/s are being transmitted, the total transmitted bit rate is MR bits/s.
The demodulator takes on the challenging task of distinguishing between each one of the 2M possible amplitudes and phases of the incoming RF carrier. As one could imagine, any distortion in the...

...to properly detect these signals. In a cable-TV system, the channel distortions arise from signal reflections caused by impedance mismatches in the cable or poorly designed television tuners whose impedance...

63/3,K/6 (Item 2 from file: 148)
DIALOG(R)File 148:IAC Trade & Industry Database

07590435 SUPPLIER NUMBER: 16402442 (USE FORMAT 7 OR 9 FOR FULL TEXT)
IF ICs offer 3 V operation for digital and analog cellular standards.
(AD607 and AD608 models from Analog Devices Inc.) (Cover Story)
Microwave Journal, v37, n10, p144(3)
Oct, 1994
DOCUMENT TYPE: Cover Story ISSN: 0192-6225 LANGUAGE: ENGLISH
RECORD TYPE: FULLTEXT
WORD COUNT: 1432 LINE COUNT: 00102

... from the master clock at 13 MHz, which is also the clock for the digital signal processor IC. The first filter is a low cost SAW. The second and third filters are simple, low cost two-pole LC filters . Both parts operate from a 3 V supply.

AD608 Design

Unlike the AD607, the AD608...

...IF. A schematic diagram of the AD 608 is shown in Figure 2. When a receiver uses a limiting IF, the limited output is de - modulated to recover the phase information and the RSSI output is used to recbver the amplitude information. Linear and limiting IF subsystems are used for GSM, PHS, TDMA and other QPSK...

63/3,K/7 (Item 3 from file: 148)
DIALOG(R)File 148:IAC Trade & Industry Database

07299598 SUPPLIER NUMBER: 15508475 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Speed and smarts pave the way for a new breed of comm chips. (high-speed technologies and new signal-processing techniques) (part of a preview of the 1994 Custom Integrated Circuits Conference)
Goldberg, Lee
Electronic Design, v42, n9, p70(5)
May 2, 1994
ISSN: 0013-4872 LANGUAGE: ENGLISH RECORD TYPE: FULLTEXT; ABSTRACT
WORD COUNT: 2678 LINE COUNT: 00220

... feedforward auto-bias adjustment to achieve both wide dynamic range and extremely high sensitivity. Conventional receivers with AGC and automatic offset control circuits typically require external passive components. To eliminate them, the team developed a receiver that employs a multistage, feedforward system to control gain and bias (Fig. 3). The receiver requires no peak-hold capacitors or other external elements.

Reducing component count by substituting digital processing for analog filtering is another growing trend, as shown by engineers from Ashi Kasei Microsystems, San Diego, Calif., and Tokyo, Japan. They developed a low-noise integrated 455-kHz IF stage filter for advanced mobile-phone service (AMPS) cellular-telephone applications. Their design employs multistage, repeating AGC and filter blocks instead of conventional ceramic filters. The first chip to apply this filtering technique demonstrated a dynamic range of 64 dB with a signal -to-(noise + distortion) ratio of 40 dB.

Parts count and power are being further reduced as the modulation and demodulation of signals is migrating to the digital domain. Designers at Siemens A.G., Munich, Germany, describe a single-chip, 60-Mbit/s quadrature- amplitude -modulation (QAM) demodulator. Implemented in 1-[micro]m CMOS, the chip performs all digital baseband processing for QAM, including carrier recovery and adjustment of bias functions in its input ADCs.

Further integration of IF functions through mixed- signal processing was demonstrated in a single-chip implementation of the quadrature phase-shift-keying (QPSK) modulation...

63/3,K/8 (Item 4 from file: 148)
DIALOG(R)File 148:IAC Trade & Industry Database

05574804 SUPPLIER NUMBER: 11844739 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Mixers hold key to unlock enhanced system performance; although it is often taken for granted, the mixer has the ability to define the limits of system performance.
Mruz, John
Microwaves & RF, v30, n12, p96(4)
Dec, 1991
ISSN: 0745-2993 LANGUAGE: ENGLISH RECORD TYPE: FULLTEXT
WORD COUNT: 1656 LINE COUNT: 00128

... components are called intermodulation products.
For transmitter applications, which require frequency upconversion, a low-frequency signal is first processed (or modulated) and then translated through a mixer to a higher microwave frequency. It is usually then amplified and transmitted. In receiver applications, which require frequency downconversion, the received signal is downconverted through a mixer to a lower frequency at which the information is extracted (demodulated).
DOWNCONVERSION
As an example of frequency downconversion, a mixer (Fig. 1) converts a 6.2-GHz input signal to an intermediate frequency (IF) of 1.1 GHz by mixing it with an internally-generated 5.1-GHz LO signal . A filter that passes the IF and rejects all other frequencies is attached to the mixer output...

63/3,K/9 (Item 5 from file: 148)
DIALOG(R)File 148:IAC Trade & Industry Database

04817619 SUPPLIER NUMBER: 08817540 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Design a DSP-based speech-transmission system. (simulation enables designers to explore DSP algorithms and hardware implementation) (technical)
Eo, Soo Kwan; Ma, Chingwo; Jang, Incheol
Electronic Design, v38, n16, p85(5)
August 23, 1990
DOCUMENT TYPE: technical ISSN: 0013-4872 LANGUAGE: ENGLISH
RECORD TYPE: FULLTEXT
WORD COUNT: 2229 LINE COUNT: 00190

... SPW library elements, and modem simulation using real-world data. Besides having an algorithmically proven signal flow, designers are provided with test points and the expected signals. Consequently, when the design...

...system architecture is best understood by visualizing the source encoding-decoding and the 9600-bps modem separately (Fig. 1). Block names are hierarchical blocks developed within the SPW from communications library functions. An adaptive delta modulator encodes the instantaneous values of the speech waveform for digital transmission through the modem . The 9600-bps modem block contains a 16 quadrature amplitude modulation (16 QAM) receiver with a linear adaptive equalizer (LAE), a model of a 3002/C4-channel telephone line, and a 16- QAM transmitter that produces a modulated , encoded, baseband signal. The LAE's output is the encoded voice signal. To recover the original voice signal, an adaptive delta demodulator decodes the instantaneous values to reproduce the voice signal. This signal is then filtered through a lowpass infinite-impulse-response (IIR) filter .
The modeled voice signal "Mary had a little lamb" was captured at

a sampling rate of 8 kHz through a GPIB interface with a Tektronix 11401 digital oscilloscope. This digitized signal was up-sampled to 9.6 kHz using interpolation and decimation filtering, functions that are contained within the source- signal block.

Delta modulation (DM) is a special case of differential pulse-coded modulation (DPCM). Here symbols from the 16-QAM signal constellation. For example, the first four symbols in the 16- QAM signal constellation are (3,3), (3,1), (1,3), (1,1). Therefore, a training sequence from...

...be (3,3).

The channel model's specification is defined by the 3002/C4 channel amplitude and delay characteristics. A time-domain filter in the SPW library performs this modeling. It reads magnitude and delay at each frequency from the input file. The filter then interpolates the data before running an inverse fast Fourier transform to output a time-domain impulse-response curve to fit the specification.

In the receiver , Hilbert transformation is applied before the coherent demodulation shifts the signal back to the baseband region. The Hilbert transformation retrieves the imaginary signal component from the transmitted signal for processing in the complex domain. Because of the linear distortion in the 3002/C4 channel, the received signal suffers severe inter-symbol-interference (ISI). Compensation for this channel imperfection is performed in the...

...coefficients are converged using training sequences within several hundred symbol periods (by monitoring the error signal from the LAE), the adaptive LAE's input is switched into the real data line...

```
?pause
>>> PAUSE started.
?
```

```
File   2:INSPEC 1969-1996/Sep W2
         (c) 1996 Institution of Electrical Engineers
File   6:NTIS 64-1996/Oct W3
         Comp & dist by NTIS, Intl Copyright All Rights Res
File   8:Ei Compendex(R) 1970-1996/Oct W1
         (c) 1996 Engineering Info. Inc.
File  77:Conference Papers Index 1973-1996/Sep
         (c) 1996 Cambridge Sci Abs
File  94:JICST-EPlus 1985-1996/Aug W3
         (c) 1996 Japan Info Center of Sci & Tech
File  99:Wilson Appl. Sci & Tech Abs 1983-1996/Jul
         (c) 1996 The HW Wilson Co.
File 108:Aerospace Database 1962-1996/Jul
         (c) 1996 AIAA
File 144:Pascal 1973-1996/Aug
         (c) 1996 INIST/CNRS
File 434:SciSearch(R) Cited Ref Sci 1974-1996/Aug W3
         (c) 1996 Inst for Sci Info
?ds

Set     Items   Description
S1     609135   VOICE OR AUDIO OR ANALOG?
S2      22160   S1(2N)SIGNAL?
S3    2482630   COMPRESS? OR REDUC? OR REDN?
S4      10113   SUBCHANNEL? OR SUBCIRCUIT? OR SUBFACILIT? OR SUBLINE? OR S-
             UBPATH? OR SUBLINK? OR SUB()(CHANNEL? OR CIRCUIT? OR FACILIT?
             OR LINE? OR PATH? OR LINK?)
S5    2705663   TIME
S6       3934   SINGLE(1W)(SIDEBAND? OR SIDE()BAND? ?)
S7          0   S2(4N)S3 AND S4 AND S5
S8          0   S7 AND S6
S9          3   S1(4N)S3 AND S4 AND S5
S10         0   S9 AND S6
S11         2   RD S9 (unique items)
?o
t11/6/all

 11/6/1     (Item 1 from file: 2)
02789561    INSPEC Abstract Number: B87003082
 Title:  Time  slotted cellular systems


 11/6/2     (Item 1 from file: 434)
13009853   Genuine Article#: NH460   Number of References: 41
Title: BIOCHEMICAL-STUDIES ON PT523, A POTENT NONPOLYGLUTAMATABLE
    ANTIFOLATE, IN CULTURED-CELLS  (Abstract Available)
?t11/7/1

 11/7/1     (Item 1 from file: 2)
DIALOG(R)File   2:INSPEC
(c) 1996 Institution of Electrical Engineers. All rts. reserv.

02789561   INSPEC Abstract Number: B87003082
 Title:  Time  slotted cellular systems
  Author(s): Cavers, J.K.; Walker, M.
  Author Affiliation: Simon Fraser Univ., Burnaby, BC, Canada
  Conference  Title: 36th  IEEE  Vehicular  Technology  Conference  (Cat.
No.86CH2308-5)     p.202-7
  Publisher: IEEE, New York, NY, USA
  Publication Date: 1986    Country of Publication: USA    455 pp.
```

U.S. Copyright Clearance Center Code: CH2308-5/86/0000-0202$01.00
Conference Sponsor: IEEE
Conference Date: 20-22 May 1986 Conference Location: Dallas, TX, USA
Language: English Document Type: Conference Paper (PA)
Treatment: Practical (P)

Abstract: A system is proposed that can carry 5 simultaneous speakers on a single standard channel. It is based on two technological trends of recent years: faster mobile data transmission and cheaper, reasonable quality compressed digital voice . By transmitting segments of speech in high-speed bursts, a single channel is time -division multiplexed among several mobiles. With several subchannels available, it becomes possible to lay out a cellular system based on reuse of time slots, rather than frequencies. The proposed system is shown to need a cluster size of 9, for a fully deployed hexagonal array. With only two operating frequencies, such a system can cover an unlimited geographic area with one subchannel per cell. For linear arrays, as along a highway, the system can be extended indefinitely with only a single operating frequency. (7 Refs)
?

```
Set     Items    Description
S1     609135    VOICE OR AUDIO OR ANALOG?

S3    2482630    COMPRESS? OR REDUC? OR REDN?
S4      10113    SUBCHANNEL? OR SUBCIRCUIT? OR SUBFACILIT? OR SUBLINE? OR S-
                 UBPATH? OR SUBLINK? OR SUB()(CHANNEL? OR CIRCUIT? OR FACILIT?
                 OR LINE? OR PATH? OR LINK?)
S5    2705663    TIME

S12        11    (BANDWID? OR BAND(1N)WIDTH)(4N)S3 AND S4
S13         0    (VOCOD? OR COMPANDOR?)(4N)S3 AND S4
S14         9    RD S12 (unique items)
S15         1    S14 AND S1
S16         1    S14 AND S5
S17         2    S15 OR S16
S18         2    RD S17 (unique items)
?
t18/7/all
```

18/7/1 (Item 1 from file: 2)
DIALOG(R)File 2:INSPEC

00088459 INSPEC Abstract Number: B70004957
Title: An effective bandwidth reduction for communication use in a VHF/UHF satellite link
Author(s): Weber, M.
Journal: IEEE Transactions on Audio and Electroacoustics vol.au-17, no.3 p.220-2
Publication Date: Sept. 1969 Country of Publication: USA
CODEN: ITADAS ISSN: 0018-9278
Language: English Document Type: Journal Paper (JP)
Abstract: A new modulation system is described by which two voice and three additional teletype subchannels are to be interlaced within a frequency range of a usual standard phone channel. In general, economical aspects govern an effective bandwidth reduction for signal communication. This becomes extremely valid for all relatively narrow-band radio links via satellites in the VHF/UHF range. With respect to a minimum of material expense, a compact band modulation system was successfully developed and tested in a satellite link. An adequately high intelligibility could be retained at the cost of some fidelity in voice communication.

18/7/2 (Item 1 from file: 108)
DIALOG(R)File 108:Aerospace Database

01461147 A84-15431
Satellite link communication protocols
PUJOLLE, G. (Paris VI, Universite, Paris, France); SPANIOL, O. (Frankfurt, Universitaet, Frankfurt am Main, West Germany)
IN: Data networks with satellites; Proceedings of the Working Conference, Cologne, West Germany, September 20, 21, 1982 (A84-15429 04-32). Berlin, Springer-Verlag, 1983, p. 37-52.
1983 9 REFS.
LANGUAGE: English
COUNTRY OF ORIGIN: France COUNTRY OF PUBLICATION: Germany
DOCUMENT TYPE: CONFERENCE PAPER
DOCUMENTS AVAILABLE FROM AIAA Technical Library

JOURNAL ANNOUNCEMENT: IAA8404

The restrictions imposed by satellite-link factors on high-level data-link control (HDLC) systems are characterized, and a new class of link protocols is presented and evaluated analytically. HDLC frame-checking, error-recovery, and window-size procedures are reviewed for the balanced operational mode and shown to significantly impair performance when HDLC is used in high-bandwidth satellite systems. 'Virtual subchannel ' protocols of static, dynamic, and priority (mixed static-dynamic) type, based on a time -slot subdivision of the channel to reduce bandwidth , increase window size, and permit the coexistence of multiple rejections, are introduced. Block diagrams of the different types are provided. The throughput behavior of the static-virtual- subchannel found that the number of subchannels must be strongly limited to control interblocking risk and error-detection time . (T.K.)

SOURCE OF ABSTRACT/SUBFILE: AIAA

?

S19 7 S14 NOT S18
?e
t19/7/all

19/7/1 (Item 1 from file: 2)
DIALOG(R)File 2:INSPEC

5185252 INSPEC Abstract Number: B9603-6250-031
Title: Performance of a coded multicarrier DS-CDMA system in multipath fading channels
Author(s): Qingxin Chen; Sousa, E.S.; Pasupathy, S.
Author Affiliation: Dept. of Electr. & Comput. Eng., Toronto Univ., Ont., Canada
Journal: Wireless Personal Communications vol.2, no.1-2 p.167-83
Publisher: Kluwer Academic Publishers,
Publication Date: 1995 Country of Publication: Netherlands
CODEN: WPCOFW ISSN: 0929-6212
SICI: 0929-6212(1995)2:1/2L.167:PCMC;1-3
Material Identity Number: D225-96001
U.S. Copyright Clearance Center Code: 0929-6212/95/$8.00
Language: English Document Type: Journal Paper (JP)
Treatment: Theoretical (T)
Abstract: Multicarrier DS-CDMA has been considered as an effective scheme for reducing multiple access interference in quasi-synchronous transmission. The scheme allows the reduction of multiple access interference by transferring the orthogonality property of the signals into the frequency domain where the orthogonality property is robust to relative chip offsets between the spreading codes of the various users. However in multipath channels, the multicarrier technique results in frequency non-selective fading in the sub - channels , due to the narrower bandwidth , hence a reduction of the capability of the spread spectrum signal to mitigate the effect of multipath propagation. We consider the use of a Reed-Muller code with soft decision decoding to regain the corresponding loss in performance, and compare the resulting system with a single carrier DS-CDMA system. The effect of system parameters such as the number of sub - channels is investigated through numerical calculation and simulation, from which a number of system design criteria are arrived at. (23 Refs)

19/7/2 (Item 2 from file: 2)
DIALOG(R)File 2:INSPEC

4446068 INSPEC Abstract Number: B9308-5210C-044
Title: ISI-reduced modulation over a fading multipath channel
Author(s): Khandani, A.K.; Kabal, P.
Author Affiliation: Dept. of Electr. Eng., McGill Univ., Montreal, Que., Canada
Conference Title: 1st International Conference on Universal Personal Communications, ICUPC '92 Proceedings (Cat. No.92TH0434-1) p.11.02/1-5
Publisher: IEEE, New York, NY, USA
Publication Date: 1992 Country of Publication: USA xxx+442 pp.
ISBN: 0 7803 0591 4
U.S. Copyright Clearance Center Code: 0 7803 0591 4/92/0000-0288$3.00
Conference Sponsor: IEEE
Conference Date: 29 Sept.-1 Oct. 1992 Conference Location: Dallas, TX, USA

Language: English   Document Type: Conference Paper (PA)
Treatment: Theoretical (T)
Abstract: The idea of using the channel eigenvectors as the basis for a block based signaling scheme over a fading multipath channel is introduced. This basis minimizes the product of the average fading attenuations along different dimensions. The ISI from the preceding blocks (intra-block ISI) is modeled by an additive Gaussian noise. To reduce the effect of the intra-block ISI, a number of zeros are transmitted between successive blocks. The number of zeros is optimized to minimize the average probability of error. As the transmission of zeros reduces the bandwidth efficiency, this optimization procedure is more useful for lower bit rates. By applying quadrature amplitude modulation (QAM) to each dimension, a set of two-dimensional subchannels with unequal fadings is obtained. A coherent M-PSK constellation is employed over each QAM subchannel . The authors propose two methods to distribute the rate and energy between the subchannels . (5 Refs)

19/7/3   (Item 3 from file: 2)
DIALOG(R)File 2:INSPEC

03983871 INSPEC Abstract Number: B91065525, C91063955
Title: Digital HDTV compression using parallel motion-compensated transform coders
Author(s): Hang, H.-M.; Leonardi, R.; Haskell, B.G.; Schmidt, R.L.; Bheda, H.; Othmer, J.H.
Author Affiliation: AT&T Bell Lab., Holmdel, NJ, USA
Journal: IEEE Transactions on Circuits and Systems for Video Technology vol.1, no.2 p.210-21
Publication Date: June 1991   Country of Publication: USA
CODEN: ITCTEM ISSN: 1051-8215
U.S. Copyright Clearance Center Code: 1051-8215/91/0600-0210$01.00
Language: English   Document Type: Journal Paper (JP)
Treatment: Practical (P); Theoretical (T)
Abstract: The authors suggest a parallel processing structure using the proposed international standard for visual telephony (CCITT P*64 kbs standard) as processing elements, to compress digital high definition television (HDTV) pictures. The basic idea is to partition an HDTV picture, in space or in frequency, into smaller sub-pictures and then compress each sub-picture using a CCITT P*64 kbs coder. This seems to be a cost-effective solution to the HDTV hardware. Since each sub-picture is processed by an independent coder, without coordination these coded sub-pictures may have unequal picture quality. To maintain a uniform quality HDTV picture, the following two issues are studied: sub - channel control strategy (bits allocated to each sub-picture); and quantization and buffer control strategy for individual sub-picture coders. Algorithms to resolve these problems and their computer simulations are presented. (13 Refs)

19/7/4   (Item 1 from file: 8)
DIALOG(R)File 8:Ei Compendex(R)

04336367 E.I. No: EIP96023018214
Title: Performance of a coded multi-carrier DS-CDMA system in multi-path fading channels
Author: Chen, Qingxin; Sousa, Elvino S.; Pasupathy, Subbarayan
Corporate Source: Univ of Toronto, Toronto, Ont, Can
Source: Wireless Personal Communications v 2 n 1-2 1995. p 167-183

Publication Year: 1995
CODEN: WPCOFW ISSN: 0929-6212
Language: English
Document Type: JA; (Journal Article) Treatment: G; (General Review)
Journal Announcement: 9603W4

Abstract: Multi-carrier DS-CDMA has been considered as an effective scheme for reducing multiple access interference in quasi-synchronous transmission. The scheme allows the reduction of multiple access interference by transferring the orthogonality property of the signals into the frequency domain where the orthogonality property is robust to relative chip offsets between the spreading codes of the various users. However in multi-path channels, the multi-carrier technique results in frequency non-selective fading in the sub - channels , due to the narrower bandwidth , hence a reduction of the capability of the spread spectrum signal to mitigate the effect of multi-path propagation. In this paper, we consider the use of a Reed-Muller code with soft decision decoding to regain the corresponding loss in performance, and compare the resulting system with a single carrier DS-CDMA system. The effect of system parameters such as the number of sub - channels is investigated through numerical calculation and simulation, from which a number of system design criteria are arrived at. (Author abstract) 23 Refs.

19/7/5 (Item 2 from file: 8)
DIALOG(R)File 8:Ei Compendex(R)

02764406 E.I. Monthly No: EI8907062453
Title: Vector coding for partial response channels.
Author: Kasturia, S.; Aslanis, J. T.; Cioffi, J. M.
Corporate Source: Stanford Univ, Stanford, CA, USA
Conference Title: IEEE 1988 International Symposium on Information Theory - Abstracts of papers
Conference Location: Kobe, Jpn Conference Date: 19880619
Sponsor: IEEE, Information Theory Group, New York, NY, USA; Inst of Electronics, Information & Communication Engineers of Japan, Jpn
E.I. Conference No.: 12027
Source: IEEE 1988 Int Symp on Inf Theory Abstr of Pap v 25 n 13. Publ by IEEE, New York, NY, USA. Available from IEEE Service Cent (cat n 88CH2621-1) Piscataway, NJ, USA. p 141-142
Publication Year: 1988
Language: English
Document Type: PA; (Conference Paper) Treatment: T; (Theoretical)
Journal Announcement: 8907

Abstract: Summary form only given, as follows. Trellis coding methods that increase the reliability of data transmission without increasing bandwidth requirements were introduced by G. Ungerboeck (1982) and subsequently have been developed extensively. Forney has characterized most of these codes as coset codes, and here the authors present a technique that applies coset codes to partial response channels. Simple codes based on the Z 4 lattice achieve asymptotic coding gains of 3 dB and more over maximum-likelihood sequence detection for partial response channels of the type (1 - D) (1 plus D) m. Their technique divides the channel into a set of parallel independent channels using eigenvectors of CC T where C specifies the pulse response of the channel over a finite input block. The authors use intersymbol-interference-free signaling on each subchannel ; they then allocate the information and redundancy bits across the subchannels . As with the precoding technique, this technique reuses known coset codes. The authors obtain significant improvement over precoding based codes for a low number of bits/T as they justify using

capacity arguments. Unlike the binary codes of Wolf-Ungerboeck, these codes expand the signal set with multilevel signals, thereby avoiding any data rate reduction or bandwidth expansion. 4 Refs.

19/7/6 (Item 3 from file: 8)
DIALOG(R)File 8:Ei Compendex(R)

00748895 E.I. Monthly No: EI7810072194 E.I. Yearly No: EI78013721
Title: TRIDEC SYSTEM DESIGN.
Author: Yasuda, Hiroshi; Kawanishi, Hisashi; Matsuoka, Takeshi; Kanaya, Fumio
Corporate Source: Nippon Telegr & Teleph Public Corp, Musashino Electr Commun Lab, Jpn
Source: Review of the Electrical Communication Laboratories (Tokyo) v 25 n 11-12 Nov-Dec 1977 p 1337-1346
Publication Year: 1977
CODEN: RELTAN ISSN: 0418-6338
Language: ENGLISH
Journal Announcement: 7810
Abstract: Design of an interframe coding system is described which is capable of transmitting both 4 MHz videotelephone signals and NTSC color TV signals at an average rate of 6. 312 Mb/s. The coding system takes as much advantage as possible of in-frame correlation, in addition to interframe correlation (combinational difference). Variable-length word coding is effectively used to reduce the number of bits to be transmitted. In order to prevent buffer overflow, luminance and spatial resolutions are controlled. To control luminance resolution, interframe difference is multiplied by coefficient ALPHA LESS THAN EQUIVALENT TO 1, so that value approaches zero when violent motion causes the buffer to fill. Spatial resolution control is achieby using subsample and subline coding modes. 14 refs.

19/7/7 (Item 4 from file: 8)
DIALOG(R)File 8:Ei Compendex(R)

00681394 E.I. Monthly No: EI7709066017 E.I. Yearly No: EI77041688
Title: TRANSMITTING 4-MHz TV SIGNALS BY COMBINATIONAL DIFFERENCE CODING.
Author: Yasuda, Hiroshi; Kuroda, Hideo; Kawanishi, Hisashi; Kanaya, Fumio ; Hashimoto, Hideo
Corporate Source: Nippon Telegr & Teleph Public Corp, Tokyo, Jpn
Source: IEEE Trans Commun V COM-25 n 5 May 1977 p 508-516
Publication Year: 1977
CODEN: IECMBT ISSN: 0096-1965
Language: ENGLISH
Journal Announcement: 7709
Abstract: An interframe coding system is described which is capable of transmitting both 4-MHz videotelephone signals and NTSC color TV signals at an average rate of 6. 312 Mbits/s. The coding system takes as much advantage as possible of in-frame correlation, in addition to interframe correlation, by coding pel-to-pel difference of interframe difference (combinational difference). Variable-length word coding is effectively used to reduce the number of bits to be transmitted. In order to prevent buffer overflow, luminance and spatial resolutions are controlled. To control luminance resolution, interframe difference is multiplied by a coefficient ALPHA LESS THAN EQUIVALENT TO 1, so that value approaches zero when violent motion causes the buffer to fill. Though multiplication with small ALPHA

results in temporal resolution reduction in moving areas, quality degradation is not too marked. Spatial resolution control is achieved by using subsample and subline coding modes. 14 refs.
?

Author Affiliation: Columbia Univ., New York, NY, USA
Journal: Journal of Computer and System Sciences vol.38, no.1 p. 134-49
Publication Date: Feb. 1989 Country of Publication: USA
CODEN: JCSSBM ISSN: 0022-0000
U.S. Copyright Clearance Center Code: 0022-0000/89/$3.00
Language: English Document Type: Journal Paper (JP)
Treatment: Theoretical (T)

Abstract: The authors show that the following statements are equivalent: Statement 1. 3-pushdown graphs have sublinear separators. Statement 1. k- page graphs have sublinear separators. Statement 2. A one-tape nondeterministic Turing machine can simulate a two-tape machine in subquadratic time . None of the statements is known to be true or false at present. However, the authors proof of equivalence is quantitative-it relates exactly the separator size of the two kinds of graphs to the running time of the simulation in Statement 2. Using this equivalence the authors derive several graph-theoretic corollaries. There are known examples where upper bounds on graph properties imply upper bounds on computation time or space. There are other examples where lower bounds on graph properties are used to derive lower bounds on computation time in restricted settings. However, the authors' results may constitute the first example where a graph problem is shown to be equivalent to a problem in computation complexity. (17 Refs)

41/7/3 (Item 1 from file: 6)
DIALOG(R)File 6:NTIS
Comp & dist by NTIS, Intl Copyright All Rights Res. All rts. reserv.

1212312 NTIS Accession Number: NTN86-1001
Simulating Single-Event Upsets in Bipolar RAM's: This simulation technique can save much testing
(NTIS Tech Note)
National Aeronautics and Space Administration, Washington, DC.
Corp. Source Codes: 011249000
Sep 86 1p
Languages: English
Journal Announcement: GRAI8623
FOR ADDITIONAL INFORMATION: Contact: NASA Technology Transfer Div., PO Box 8757 BWI Airport, MD 21240; (301) 621-0100 ext 241. Refer to NPO-16491/TN.
NTIS Prices: Not available NTIS
Country of Publication: United States

This citation summarizes a one-page announcement of technology available for utilization. A technical paper presents the results of computer simulations of single-event upsets in bipolar integrated circuits--particularly random-access memories (RAM's). A companion paper discusses experiments to check the accuracy of the simulation. Although metal-oxide-semiconductor (MOS) integrated circuits have received more attention, bipolar chips are also susceptible to single-event upsets and will become even more so as they are made smaller. Also known as single-particle soft-error generation, this phenomenon is manifested as a bit flip in a flip-flop or a RAM cell. It occurs when an energetic heavy ion traverses a transistor or diode in a chip. The simulation uses an interactive version of SPICE (Simulation Program with Integrated Circuit Emphasis). The device and subcircuit models available in the software are used to construct a macromodel for an integrated bipolar transistor. Time-dependent current generators are placed inside the transistor macromodel to simulate charge collection from an ion track. A significant finding of the experiments is that the standard design practice of reducing the power in an unaddressed bipolar RAM cell increases the sensitivity of the cell to single-event upsets. The reason for this is that a RAM cell operating in the unaddressed, standby mode responds more slowly and allows charge to collect over a longer time period. An unaddressed cell may respond to an ion track anywhere in the bipolar transistor through charge collection at all junctions.

41/7/4 (Item 1 from file: 434)
DIALOG(R)File 434:SciSearch(R) Cited Ref Sci

13704745 Genuine Article#: QJ657 Number of References: 39
Title: MEASLES-VIRUS PERSISTENCE IN AN IMMORTALIZED MURINE MACROPHAGE CELL-LINE
Author(s): GOLDMAN MB; BUCKTHAL DJ; PICCIOTTO S; OBRYAN TA; GOLDMAN JN
Corporate Source: PENN STATE UNIV,DEPT MED/HERSHEY//PA/17033; PENN STATE UNIV,COLL MED,DEPT MICROBIOL & IMMUNOL/HERSHEY//PA/17033
Journal: VIROLOGY, 1995, V207, N1 (FEB 20), P12-22
ISSN: 0042-6822
Language: ENGLISH Document Type: ARTICLE
Abstract: Persistent infection with the Edmonston strain of measles virus (MV) has been established in IC-21 cells, an immortalized murine macrophage cell line. Persistence was established immediately without syncytia formation or cytopathic effects. MV was expressed in the majority of the cells as evidenced by immunofluorescence microscopy, flow cytometry, infectious centers assays, and limiting dilution analysis. Hemagglutinin (H) and phosphoprotein expressed in persistently infected IC-21 cells had retarded migration in SDS-PAGE gels when compared to these proteins expressed in Vero cells. H protein differences were also found between freshly infected IC-21 cells and persistently infected IC-21 cells passaged for over 2 years. Six sublines of IC-21 cells, infected at different times, have maintained these characteristics for 2 years of passage. During this time period the intensity of immunofluorescence and the number of infectious virus particles recoverable fluctuated in five of the six cell lines. In one cell line virus expression remained at a consistent high level. The ability to establish a persistent MV infection in murine macrophages allows studies using a cell important in disseminating the infection. It facilitates experiments on immunological aspects of viral immunity by enabling cell mixing experiments with histocompatible cell populations and by making available the wide array of cellular and humoral reagents in the mouse. (C) 1995 Academic Press, Inc.

41/7/5 (Item 2 from file: 434)
DIALOG(R)File 434:SciSearch(R) Cited Ref Sci

12299178 Genuine Article#: LA569 Number of References: 16
Title: MOST UNIFORM PATH PARTITIONING AND ITS USE IN IMAGE-PROCESSING
Author(s): LUCERTINI M; PERL Y; SIMEONE B
Corporate Source: UNIV ROMA LA SAPIENZA,DEPT STAT,PIAZZALE ALDO MORO 5/I-00185 ROME//ITALY/; UNIV ROMA LA SAPIENZA,DEPT STAT,PIAZZALE ALDO MORO 5/I-00185 ROME//ITALY/; NEW JERSEY INST TECHNOL,INST INTEGRATED SYST,DEPT COMP & INFORMAT SCI/NEWARK//NJ/07102; UNIV ROMA TOR VERGATA,DEPT ELECT ENGN/I-00173 ROME//ITALY/; UNIV ROMA TOR VERGATA,CTR MATEMAT VITO VOLTERRA/I-00173 ROME//ITALY/
Journal: DISCRETE APPLIED MATHEMATICS, 1993, V42, N2-3 (APR 27), P227-256

ISSN: 0166-218X
Language: ENGLISH   Document Type: ARTICLE
Abstract: Let Q be a vertex-weighted path with n vertices. For any pair (L, U) can one find a partition of Q into (a given number p of) subpaths, such that the total weight of every subpath lies between L and U?. We present linear- time algorithms for the partitioning problem for given (L, U) and an O(n2p/log n) algorithm, relying on the above procedures, for finding a partition that minimizes the difference between the largest and the smallest weight of a subpath (most uniform partitioning). Our approach combines a preprocessing procedure, which detects ''obstructions'', if any, via a sequence of vertex compressions; and a greedy procedure, which actually finds the desired partition. Path partitioning can be a useful tool in facing image degradation. In fact whenever a picture is taken or converted from one form to another, the resulting image can be affected by different types and degrees of degradation; if we have no informations on the actual degradation process that has taken place on the image (or if it is too difficult or costly to find such informations), the only way for image enhancement consists in increasing contrast and reducing noise by suitable modifications of the grey level of pixels. Finding the optimal grey scale transformation which leads to this enhancement can be formulated as the problem of partitioning into connected components a path with vertices corresponding to grey levels and vertex weights equal to the number of occurrences of the corresponding tone in the image, so that the sum of the weights of the vertices in each component is ''as constant as possible''. In addition to image processing, this problem has applications in paging, clustering and the design of communication networks.
?

Set	Items	Description
S1	609135	VOICE OR AUDIO OR ANALOG?
S30	116424	DEMODULAT? OR DE()MODULAT? OR MODEM? OR S1(3N)DIGIT?
S42	16055	S30 AND (RECEIV? OR TERMINAL?)
S44	2883	S42 AND FILTER?
S45	1401	S44 AND (MODULAT? OR QAM)
S46	383	S45 AND AMPLI?
S47	69	S46 AND SIGNAL?(2N)PROCESS?
S48	33	S47 AND S1
S49	29	RD S48 (unique items)
S50	3	S49/1996
S51	26	S49 NOT S50

51/7/1 (Item 1 from file: 2)
DIALOG(R)File 2:INSPEC

5358466 INSPEC Abstract Number: B9610-6320-006, C9610-5260-039
Title: Digital signal processing in a spaceborne radar altimeter
Author(s): Crowley, R.D.; Walker, D.M.
Author Affiliation: E-Syst. Inc., St. Petersburg, FL, USA
Conference Title: Proceedings of the 5th International Conference on Signal Processing Applications and Technology Part vol.2 p.1338-43 vol.2
Publisher: DSP Associates, Waltham, MA, USA
Publication Date: 1994 Country of Publication: USA 2 vol. 1754 pp.
Material Identity Number: XX96-01953
Conference Title: Proceedings of 5th International Conference on Signal Processing Applications and Technology
Conference Date: 18-21 Oct. 1994 Conference Location: Dallas, TX, USA
Language: English Document Type: Conference Paper (PA)
Treatment: Practical (P); Theoretical (T)
Abstract: This paper gives the design of the DSP function in a spaceborne radar altimeter. The digital signal processing function begins with the antialias filtering operation that consists of an isolation buffer amplifier, an antialias filter, and an AC-coupled A/D buffer amplifier. The output of this function drives an 8-bit A/D converter. Two identical channels are implemented to produce a complex signal. Filter requirements are eased by oversampling the data. 256 complex samples are taken of each return chirp. To make fine tracking adjustments, the samples are frequency translated via a complex multiply by a complex tone. The data then has a 256-point complex FFT performed on it. Next, the magnitude squared function is taken of each individual frequency bin. The center 128 frequencies are used for further processing. This begins by accumulating the power in 50 successive chirps to smooth the random data. The processing to this point is implemented in a 16-bit fixed point digital signal processor, and control logic is resident in a field programmable gate array (FPGA). The 50-chirp average is then used as the input to two tracking loops that adjust the receiver gain, LO chirp timing (coarse timing), and fine timing. (6 Refs)

51/7/2 (Item 2 from file: 2)

5248294 INSPEC Abstract Number: B9606-6430D-003
Title: 64/256 QAM chip-set for digital CATV application
Author(s): Ishizawa, Y.; Yanai, Y.; Nakayama, I.
Conference Title: IEEE 1995 International Conference on Consumer Electronics. Digest of Technical Papers. ICCE (Cat. No.95CH3571-9) p. 36-7
Publisher: IEEE, New York, NY, USA
Publication Date: 1995 Country of Publication: USA 430 pp.
ISBN: 0 7803 2140 5 Material Identity Number: XX95-01174
U.S. Copyright Clearance Center Code: 0 7803 2140 5/95/$3.00
Conference Title: Proceedings of International Conference on Consumer Electronics
Conference Sponsor: Consumer Electron. Soc. IEEE
Conference Date: 7-9 June 1995 Conference Location: Rosemont, IL, USA
Language: English Document Type: Conference Paper (PA)
Treatment: Applications (A); Practical (P)
Abstract: A small and cost effective 16/32/64/256 QAM receiver chip-set is developed for next generation digital CATV systems. The QAM receiver was partitioned into a two-chip set, analog and digital chip for board size reduction. Most of all the analog parts except the tuner is incorporated in one chip. The digital adaptive equalizer with about 470 k-transistors has been implemented on a 35 mm/sup 2/ die using a 0.5 mu m CMOS technology and with an optimized filter circuit. (2 Refs)

51/7/3 (Item 3 from file: 2)
DIALOG(R)File 2:INSPEC

03692448 INSPEC Abstract Number: B90057382
Title: Characteristics of land mobile earth stations for the EMSS project
Author(s): Hamamoto, N.; Suzuki, R.; Ikegami, T.; Taira, S.; Ichiyoshi, O.; Sato, N.; Tada, S.
Journal: Review of the Communications Research Laboratory vol.36, spec. issue. p.75-82
Publication Date: March 1990 Country of Publication: Japan
CODEN: TSKKED ISSN: 0914-9279
Language: Japanese Document Type: Journal Paper (JP)
Treatment: Applications (A); Practical (P); Experimental (X)
Abstract: A feasibility study on land mobile satellite communication was carried out using three kinds of mobile earth stations. These earth stations were developed after considering the potential effect of such signal propagation characteristics as multipass fading and the blockage effect on future practical applications. The first earth station employs the ACSSB (Amplitude Companded SSB) technique, which can transmit a voice signal with narrow bandwidth. The second earth station uses a digital signal processing technique applied to both QPSK modulation / demodulation and voice coding/decoding. The third earth station employs the spread spectrum (SS) system using a matched filter technique for de-spreading a receiving signal. The paper introduces the configurations of these earth stations and also reports on their performance. Additionally, an example of signal propagation characteristics in an urban area is presented. These are experimental results using the ETS-V satellite. (5 Refs)

51/7/4 (Item 4 from file: 2)

DIALOG(R)File 2:INSPEC

03568728 INSPEC Abstract Number: B90017768
Title: A mixed analog - digital secondary channel FSK modem
Author(s): Chen, C.-S.; Huang, E.; White, B.J.
Author Affiliation: Silicon Syst. Inc., Tustin, CA, USA
Conference Title: 1989 IEEE International Solid-State Circuits Conference. Digest of Technical Papers. 36th ISSCC. First Edition (Cat. No.CH2684-9) p.266-7, 363
Publisher: IEEE, New York, NY, USA
Publication Date: 1989 Country of Publication: USA 394 pp.
U.S. Copyright Clearance Center Code: 0193-6530/89/0000-0266$01.00
Conference Sponsor: IEEE; Univ. Pennsylvania
Conference Date: 15-17 Feb. 1989 Conference Location: New York, NY, USA
Language: English Document Type: Conference Paper (PA)
Treatment: Applications (A); Practical (P)
Abstract: The authors describe a fully integrated circuit that performs the functions of a 75-b/s narrowband FSK (frequency-shift-keying) modem , along with the necessary filtering functions to provide greater than 30-dB channel separation between FSK and QAM (quadrature amplitude - modulation) signals. Switched-capacity circuits are used extensively to implement the QAM filters and the FSK transmitter, and a reduced-instruction-set digital signal processor is employed to realize the FSK receiver . An important aspect of the digital filters realized on the chip is that each multiplier coefficient is optimized to have fewer than three nonzero bits in the signed-digit representation. With layout precautions and separate bonding pads for voltage supplies, the crosstalk between QAM and FSK signals and the coupling of the signal processor switching noise into analog signals are minimized. Less than 0.5 dB performance degradation of signal-to-noise ratio with 10/sup -6/ bit-error rate is introduced for 19.2-kb/s modems . The chip is fabricated with a 3- mu m, double-polysilicon, single-metal, p-well CMOS process. (2 Refs)

51/7/5 (Item 5 from file: 2)
DIALOG(R)File 2:INSPEC

03070367 INSPEC Abstract Number: B88015324
Title: The generation and demodulation of SSB signals using the phasing method. II. Signal processing for a SSB/DSB/AM-transceiver without using crystal filters
Author(s): Oppelt, R.
Journal: VHF Communications vol.19, no.3 p.130-40
Publication Date: Autumn 1987 Country of Publication: West Germany
CODEN: VHFCEB ISSN: 0177-7505
Language: English Document Type: Journal Paper (JP)
Treatment: Practical (P)
Abstract: For pt.I see ibid., vol.19, no.2, p.66 (1987). A practical realization of phasing SSB processing is described. The experimental prototype presented contains the phase-shifting networks and mixer stages for both transmitter and receiver . The single-side-band signal is capable of being switched between upper and lower sideband and the term double-side-band refers to two sidebands carrying identical modulation but having a suppressed carrier. The AM signal is, of course, the DSB signal complete with full carrier. Switching between send and receive as well as between the three modes is accomplished at audio frequency

by means of CMOS switches. (0 Refs)

51/7/6 (Item 6 from file: 2)
DIALOG(R)File 2:INSPEC

02445907 INSPEC Abstract Number: B85030775, C85024310
Title: A 1200 Bit/s QPSK full duplex modem
Author(s): Hanson, K.; Severin, W.A.; Klinkovsky, E.R.; Richardson, D.C.; Hochschild, J.R.
Author Affiliation: Texas Instrum. Inc., Houston, TX, USA
Journal: IEEE Journal of Solid-State Circuits vol.SC-19, no.6 p. 878-87
Publication Date: Dec. 1984 Country of Publication: USA
CODEN: IJSCBC ISSN: 0018-9200
U.S. Copyright Clearance Center Code: 0018-9200/84/1200-0878$01.00
Language: English Document Type: Journal Paper (JP)
Treatment: Practical (P)
Abstract: An NMOS integrated circuit is described that provides all the modulation , demodulation - filtering , and data-buffering functions for a 1200-b/s full-duplex voice -band modem . This modem is compatible with the Bell 212A modem , the Racal-Vadic 2400 modem , and the CCITT V.22 recommendation. All of the modems , options, and alternatives supported by these modems are also supported by the integrated circuit. The integrated circuit uses switched-capacitor signal processing circuit technology. This technique uses charge storage on-ratioed capacitors and allows the design of filters , amplifiers , automatic gain control elements, voltage-controlled oscillators, Hilbert filters , phase-locked loops, and a fully adaptive equalizer to be implemented on the same integrated circuit as the digital functions, which include the clock generation circuits and the transmit and receive buffers. Performance measurements show that the IC meets the bit error rate standard at 12-dB signal-to-noise ratio. (12 Refs)

51/7/7 (Item 7 from file: 2)
DIALOG(R)File 2:INSPEC

02064444 INSPEC Abstract Number: B83035180
Title: Hybrid matched filters for discrete FM signals and their noise immunity
Author(s): Zhayloobayev, N.
Journal: Elektrosvyaz vol.36, no.3 p.58 et seq.
Publication Date: March 1982 Country of Publication: USSR
CODEN: EKVZAO ISSN: 0013-5771
Translated in: Telecommunications and Radio Engineering, Part 1 (Telecommunications) vol.36, no.3 p.57-61
Publication Date: March 1982 Country of Publication: USA
CODEN: TCREAG ISSN: 0040-2508
Language: English Document Type: Journal Paper (JP)
Treatment: Theoretical (T)
Abstract: Discrete, frequency- modulated (DFM) signals are widely used in noise-immune systems for transmitting sampled data. A DFM signal can be received by a receiving device which consists of linear (analogue) matched filters (LMF). Each LMF contains N channels and an output adder which adds together the outputs of all channels. Each channel consists of a matched filter which is tuned to the frequency and phase of a specific element of the DFM signal (below such a filter

will be called an element matched filter -EMF), a delay line and a phase shifter. A receiver containing an LMF consists of RF amplifiers, mixers, heterodyne oscillators and IF amplifiers, and AGC system, an optimum signal processor with LMF, a secondary processor and a synchronization device. An optimum LMF processor, however, is optimized with respect to random noise and is quite ineffective in the presence of high-power structural (similar to the information-carrying signal), pulsed and narrowband interference (SI, PI and NBI). This disadvantage can be eliminated by replacing the LMF in such a processor by a hybrid (digital - analogue) filter (HF). The need for such a filter was discussed together with different versions of hybrid filters for DFM signals. In this paper the authors discuss the most promising version of the HF, which works effectively in the presence of random noise, SI, PI and NBI. As a first-order approximation, the noise immunities of LMF and HF processors can be estimated from changes in the output signal-to-interference ratios of these filters when signal, random noise and high-power SI, PI and NBI are present at their inputs. (5 Refs)

51/7/8 (Item 8 from file: 2)
DIALOG(R)File 2:INSPEC

01867245 INSPEC Abstract Number: B82032215
Title: Fully digital-a revolution in colour television receivers
Author(s): Knapp, K.H.
Journal: Funkschau no.24 p.95-9
Publication Date: 27 Nov. 1981 Country of Publication: West Germany
CODEN: FUSHA2 ISSN: 0016-2841
Language: German Document Type: Journal Paper (JP)
Treatment: Applications (A); General, Review (G)

Abstract: Describes a novel ITT TV receiver, operating in digital mode below the IF amplifiers, i.e. in the video- and audio stages. To achieve an economically viable solution, all applied A/D converters operate at an optimal minimum bit rate. The central processor is based on the IC type 8049, and controls the video codec and video processor, with digital delay and adder- and multiplier circuits for luminance and chrominance separation and amplification, with a clock rate of 18 MHz. A separate digital processor provides sync control- and lock-signals for both time bases. While digital video processing requires a large bandwidth, the audio system with the stereo TV sound demands high resolution, typically 14 bit. Pulse width modulation with either class-D or linear amplifiers after lowpass- filtering is applied. Block diagrams of video- and audio digital sections are shown and commented upon. The whole concept employs 6 special ICs and saves about 360 individual components; photos of a conventional and of the new chassis demonstrate how 'empty' the latter is. Of main interest are speculations on future prospects of digital TV broadcasting, completely digitalized receivers, requiring more chip integration, novel test techniques and service approaches. Picture reproduction would greatly benefit by higher field frequency, (75 Hz are proposed), freedom from line flicker and suppression of ghost images. (1 Refs)

51/7/9 (Item 9 from file: 2)
DIALOG(R)File 2:INSPEC

00478778 INSPEC Abstract Number: B73005774, C73003278

Title: A multimode digital processing transceiver
Author(s): Abend, K.; Gumacos, C.
Author Affiliation: Philco-Ford Corp., Willow Grove, PA, USA
Conference Title: Proceedings of the IEEE 1972 International Conference on Communications p.44-26/6 pp.
Editor(s): Salati, O.M.; Weinrich, A.W.
Publisher: IEEE, New York, NY, USA
Publication Date: 1972 Country of Publication: USA xviii+1154 pp.
Conference Sponsor: IEEE
Conference Date: 19-21 June 1972 Conference Location: Philadelphia, PA, USA
Language: English Document Type: Conference Paper (PA)
Treatment: Experimental (X)
Abstract: This paper reports on the results of a series of investigations related to the use of a real-time digital processor to perform all tactical transceiver functions of modulation , demodulation , frequency synthesis, heterodyning, and filtering . A breadboard Digital Transceiver that was built and tested is described. By switching between several programs the processor becomes either a receiver or a transmitter; for either voice or digital data; via amplitude , single-sideband, phase, or frequency modulation . Algorithms were developed which reduce the complexity of the processor and result in a minimum-complexity design for the transceiver. The resulting Multimode Transceiver system configuration including filtering , resampling, frequency translation, bandpass sampling, and demodulation is described. (3 Refs)

51/7/10 (Item 1 from file: 6)
DIALOG(R)File 6:NTIS

1759853 NTIS Accession Number: N94-21567/0/XAB
QPPM Receiver for Free-Space Laser Communications
Budinger, J. M. ; Mohamed, J. H. ; Nagy, L. A. ; Lizanich, P. J. ; Mortensen, D. J.
National Aeronautics and Space Administration, Cleveland, OH. Lewis Research Center.
Corp. Source Codes: 019039001; ND315753
Report No.: NAS 1.15:106424; E-8265; NASA-TM-106424; AIAA PAPER 94-1160
Jan 94 12p
Languages: English
Journal Announcement: GRAI9410; STAR3205
Proposed for Presentation at the 15TH International Communications Satellite Systems Conference, San Diego, Ca, 28 Feb. - 3 Mar. 1994; Sponsored by Aiaa.
NTIS Prices: PC A03/MF A01
Country of Publication: United States
Contract No.: RTOP 506-72-21
A prototype receiver developed at NASA Lewis Research Center for direct detection and demodulation of quaternary pulse position modulated (QPPM) optical carriers is described. The receiver enables dual-channel communications at 325-Megabits per second (Mbps) per channel. The optical components of the prototype receiver are briefly described. The electronic components, comprising the analog signal conditioning, slot clock recovery, matched filter and maximum likelihood data recovery circuits are described in more detail. A novel digital symbol clock recovery technique is presented as an alternative to conventional analog methods. Simulated link degradations including noise and pointing-error induced amplitude variations are applied. The bit-error-rate performance of the electronic

portion of the prototype receiver under varying optical signal-to-noise power ratios is found to be within 1.5-dB of theory. Implementation of the receiver as a hybrid of analog and digital application specific integrated circuits is planned.

51/7/11 (Item 2 from file: 6)
DIALOG(R)File 6:NTIS

1678718 NTIS Accession Number: AD-A260 811/5/XAB
I/Q Demodulation of Radar Signals with Calibration and Filtering
Lee, J. P.
Defence Research Establishment, Ottawa (Ontario).
Corp. Source Codes: 054855000; 404576
Report No.: DREO-1119
Dec 91 67p
Languages: English
Journal Announcement: GRAI9313
NTIS Prices: PC A04/MF A01
Country of Publication: Canada

A simple in-phase/quadrature (I/Q) demodulator architecture which can measure the amplitude, phase and instantaneous frequency of radar signals is examined in this report. This architecture has a potential for radar ESM applications where wide instantaneous bandwidth and simple algorithms for extracting the modulation characteristics of radar signals are required. This I/Q architecture can meet the requirement by splitting an incoming signal into its in-phase and quadrature components using analog circuitries. However in practice, there are amplitude and phase imbalances between the two components and DC offsets, which can introduce large systematic errors to the measurement. In this report, we present novel techniques which can greatly reduce the systematic errors and improve the accuracy of the measurement. A time-domain analysis on the systematic errors is given. A calibration technique which can be used to correct for the imbalances and offsets is discussed. The effect of noise on the accuracy of the measurement is also examined. Imbalance errors and DC offsets of I/Q networks are measured and analyzed. Finally, a post-processing technique employing moving averages, which is shown to be effective for improving the output signal-to-noise ratio and reducing systematic errors, is also presented....I/Q Demodulation, Radar ESM, Digital receiver, Radar signal parameter measurement.

51/7/12 (Item 3 from file: 6)
DIALOG(R)File 6:NTIS

747956 NTIS Accession Number: AD-D006 515/1
Two-Bit A/D Conversion Apparatus without a Signal Derived Automatic Gain Control
(Patent)
Covitt, Arthur L.
Department of the Air Force, Washington, DC.
Corp. Source Codes: 000260000
Report No.: PAT-APPL-768 812; PATENT-4 168 470
Filed 15 Feb 77 patented 18 Sep 79 5p
Languages: English Document Type: Patent
Journal Announcement: GRAI8006
Supersedes PAT-APPL-768 812-77, AD-D003 527.
This Government-owned invention available for U.S. licensing and,

possibly, for foreign licensing. Copy of patent available Commissioner of Patents, Washington, DC 20231 $0.50.
NTIS Prices: Not available NTIS
Country of Publication: United States
The present invention utilizes a receiver demodulator to provide an accurate analog to digital conversion of a phase modulated carrier without a signal derived automatic gain control (AGC). The two-bit A/D conversion apparatus utilizes a pair of rectifier units to resolve the carrier baseband residue into a pair of quadrature channel voltages. The quadrature channel voltages are continuously applied as control signals to the A/D converters to select the instantaneously appropriate binary output signal from each channel. For any signal-to-noise ratio input exceeding an acceptable minimum, the output is independent of input signal amplitude and therefore the A/D converter performs a function equivalent to an ideal instantaneous automatic gain control (IAGC). The digitized outputs are sequentially sampled in a decommuator unit and applied respectively to a pair of digital matched filters for digital signal processing appropriate to the particular type of phase modulation used in the present embodiment.

51/7/13 (Item 1 from file: 8)
DIALOG(R)File 8:Ei Compendex(R)

04242359 E.I. No: EIP95092844447
Title: Base band processing IC for analogue cordless telephones
Author: Okanobu, Taiwa; Hareyama, Nobuo; Yokoyama, Hiroshi; Yamazaki, Daisuke; Kamikubo, Yasunobu; Dapat, George II
Corporate Source: Sony Corp, Tokyo, Jpn
Conference Title: Proceedings of the 1995 IEEE International Conference on Consumer Electronics
Conference Location: Rosemont, IL, USA Conference Date: 19950607-19950609
E.I. Conference No.: 43508
Source: Digest of Technical Papers - IEEE International Conference on Consumer Electronics 1995. IEEE, Piscataway, NJ, USA,95CH3571-9. p 338-339
Publication Year: 1995
CODEN: DTPEEL ISSN: 0747-668X
Language: English
Document Type: CA; (Conference Article) Treatment: A; (Applications); G ; (General Review)
Journal Announcement: 9510W5
Abstract: We have designed an audio and data processing circuit for cordless telephone that contains a Scrambler, a Modem , and amplifiers for microphone and receiver , and that can be fabricated into a single chip IC. (Author abstract)

51/7/14 (Item 2 from file: 8)
DIALOG(R)File 8:Ei Compendex(R)

04242229 E.I. No: EIP95092844317
Title: 64/256 QAM chi-set for digital CATV application
Author: Ishizawa, Yoshiro; Yanai, Yoshimasa; Nakayama, Isao
Corporate Source: ULSI Systems Development Lab, Kawasaki, Jpn
Conference Title: Proceedings of the 1995 IEEE International Conference on Consumer Electronics
Conference Location: Rosemont, IL, USA Conference Date: 19950607-19950609

E.I. Conference No.: 43508
Source: Digest of Technical Papers - IEEE International Conference on Consumer Electronics 1995. IEEE, Piscataway, NJ, USA,95CH3571-9. p 36-37
Publication Year: 1995
CODEN: DTPEEL ISSN: 0747-668X
Language: English
Document Type: CA; (Conference Article) Treatment: T; (Theoretical)
Journal Announcement: 9510W5
Abstract: A small and cost effective 16/32/64/256 QAM receiver chip-set is developed for next generation digital CATV systems. The QAM receiver was partitioned into a two-chip set, analog and digital chip for the board size reduction. Most of all analog parts except tuner is incorporated in one chip. The digital adaptive equalizer with about 470k-transistors has been implemented on a 35mm 2 die using a 0.5 mu m CMOS technology and with the optimized filter circuit. (Author abstract) 2 Refs.

51/7/15 (Item 3 from file: 8)
DIALOG(R)File 8:Ei Compendex(R)
(c) 1996 Engineering Info. Inc. All rts. reserv.

04199137 E.I. No: EIP95072764752
Title: Asymptotic clipping noise distribution and its impact on M-ary QAM transmission over optical fiber
Author: Shi, Qun
Corporate Source: Panasonic Technologies, Inc, Princeton, NJ, USA
Source: IEEE Transactions on Communications v 43 n 6 Jun 1995. p 2077-2084
Publication Year: 1995
CODEN: IECMBT ISSN: 0090-6778
Language: English
Document Type: JA; (Journal Article) Treatment: T; (Theoretical)
Journal Announcement: 9509W1
Abstract: Performance analysis for a hybrid subcarrier-multiplexed (SCM) AM/ QAM transmission over an optical fiber is presented. It is shown that the bit-error-rate (BER) of M-ary QAM in such hybrid systems can be significantly affected due to occasional laser 'clipping' of the SCM signal. Here we analytically determine the asymptotic distribution of the clipping noise by modeling it as a Poisson impulse train. The BER of M-ary QAM is then evaluated in terms of optical modulation indices of M-ary QAM and AM signals which in turn specify signal-to-noise ratio (SNR), impulsive index (clipping index) of the clipping noise, and power ratio of the Gaussian noise to the clipping noise. Numerical examples are given and compared with experimental data with reasonably good agreement for small SNR's. The results have application for estimating the BER's of digital signals for SCM analog / digital transmission over an optical fiber and for employing appropriate error correction codes and/or optimum receiver design for such environments. (Author abstract) 23 Refs.

51/7/16 (Item 4 from file: 8)
DIALOG(R)File 8:Ei Compendex(R)
(c) 1996 Engineering Info. Inc. All rts. reserv.

04110049 E.I. No: EIP95032621784
Title: Wideband RF downconverters for digital telemetry receivers
Author: Bancroft, Douglas
Corporate Source: Naval Command, Control, and Ocean Surveillance Cent,

Warminster, PA, USA
Conference Title: Proceedings of the 1994 Tactical Communications Conference
Conference Location: Fort Wayne, IN, USA Conference Date: 19940510-19940512
Sponsor: ARPA; IEEE; AFCEA
E.I. Conference No.: 42621
Source: Tactical Communications Conference - Proceedings v 1 1994. IEEE, Piscataway, NJ, USA, 94TH0678-3. 265p
Publication Year: 1994
CODEN: 001965
Language: English
Document Type: CA; (Conference Article) Treatment: G; (General Review); T; (Theoretical)
Journal Announcement: 9505W2
Abstract: The development of extremely high sampling rate, high dynamic range A/D converters has resulted in the feasibility of a mostly digital telemetry receiver . The Naval Command and Control and Ocean Surveillance Center, under Office of Navy Research funding, is developing an Advanced ASW Receiver which implements RF channel filtering , null steering, direction finding and information demodulation in a high speed Digital Signal Processor . This paper describes the analysis conducted for the analog RF circuitry required to downconvert the multi-channel VHF sonobuoy band to baseband compatible with current technology A/D counts. Performance parameters critical to the system design include good spurious free performance, sharp band edge filtering , and an AGC scheme which permits phase and amplitude match between antenna input channels. The paper discusses the analysis tradeoffs performed to arrive at the downconverter design, and includes a discussion of the components selected for the receiver hardware implementation as well as the component mounting and shielding techniques. (Author abstract)

51/7/17 (Item 5 from file: 8)
DIALOG(R)File 8:Ei Compendex(R)

04082942 E.I. No: EIP95022584953
Title: Timing and carrier recovery techniques in DSP based digital receivers
Author: Sheth, Shruti; Harris, Fred
Corporate Source: San Diego State Univ, San Diego, CA, USA
Conference Title: Proceedings of the 1994 IEEE MILCOM. Part 2 (of 3)
Conference Location: Long Branch, NJ, USA Conference Date: 19941002-19941005
Sponsor: IEEE
E.I. Conference No.: 42512
Source: IEEE MILCOM v 2 1994. IEEE, Piscataway, NJ, USA, 94CH3400-9. p 639-643
Publication Year: 1994
CODEN: 001918
Language: English
Document Type: CA; (Conference Article) Treatment: A; (Applications); T ; (Theoretical)
Journal Announcement: 9504W4
Abstract: In conventional receivers , carrier recovery and timing recovery are performed in the analog domain by controlling the frequency and phase of voltage controlled oscillators (VCO) in their respectivephase locked loop (PLL). When the control signal for these loops are generated in the sampled data domain by DSP techniques the digital

samples must be brought to the analog domain by a pair of digital -to- analog converters (DAC). It is more cost effective to perform the entire signal processing function of the PLL in the digital domain and avoid the cost of the DAC and analog smoothing filter in the processing loops. In the full DSP implementation the receiver performs an initial complex down conversion with an asynchronous local oscillator set to the nominal final conversion frequency and then absorbs the residual carrier and phase uncertainty by data dependent control of a digital complex rotator. In a similar fashion sample timing is performed by the sampling the input signal with an asynchronous sampling clock operating at nominally twice the symbol rate and then absorbs residual frequency and phase of the sampling clock by resampling the data with a polyphase filter bank. (Author abstract) 4 Refs.

51/7/18 (Item 6 from file: 8)
DIALOG(R)File 8:Ei Compendex(R)

03615516 E.I. No: EIP93020706938
Title: Method of designing adaptive group digital multichannel 'two-of-six' code receivers for PCM signals
Author: Braynina, I.S.
Source: Telecommunications and Radio Engineering (English translation of Elektrosvyaz and Radiotekhnika) v 47 n 2 Feb 1992. p 8-14
Publication Year: 1992
CODEN: TCREAG ISSN: 0040-2508
Language: English
Document Type: JA; (Journal Article) Treatment: T; (Theoretical); A; (Applications)
Journal Announcement: 9306W2
Abstract: In automatic telephone communications dialing signals are transmitted in a 'two-of-six' multifrequency code. The number information is generated in the tone frequency spectrum in combinations of two frequencies from the series 700, 900, 1100, 1300, 1500 and 1700 Hz. Analog number dialing signals are converted into group digital streams in multichannel pulse-code modulation (PCM) systems. The initial information must be extracted directly from the digital streams by a group digital receiver without preliminary conversion of the digital signals into analog signals. To improve reception noise immunity without increasing filter complexity, control of the receiving threshold in proportion to the total input signal has been used in the design of systems. Threshold reduction, however, can reduce nonlinear interference and noise protection. Based on permissible level 'skews' (amplitude range, frequency deviations from nominal values, and reception/nonreception thresholds), this paper develops a simplified version of a digital 'two-of-six' code receiver that satisfies less stringent conditions for transmission over long-distance channels, trunks and recording trunks compared with physical lines. 5 Refs.

51/7/19 (Item 7 from file: 8)
DIALOG(R)File 8:Ei Compendex(R)

01036909 E.I. Monthly No: EI8108069407 E.I. Yearly No: EI81086791
Title: ANALOG ENCODED SIGNAL TRANSMISSION USING POSITION-MODULATED LINEAR CHIRPS.
Author: Kowatsch, M.; Lafferl, J.; Seifert, F. J.
Corporate Source: Tech Univ, Vienna, Aust

Source: Ultrasonics Symposium Proceedings v 1, Boston, Mass, Nov 5-7 1980. Publ by IEEE (Cat n 80CH1602-2), Piscataway, NJ, 1980 p 18-21
Publication Year: 1980
CODEN: ULSPDT ISSN: 0090-5607
Language: ENGLISH
Journal Announcement: 8108

Abstract: An analog encoded transmission system is presented, which operates with linear FM (chirp) pulses that are position- modulated according to the sampled baseband data. Chirp duration and sampling rate are properly chosen to yield permanent overlapping of several FM pulses. To recover the baseband signal in the receiver , separation of consecutive chirps by pulse compression in a matched filter is required. Synchronization is attained by phase-locking the receiver reference quartz oscillator to the pulse position modulation (PPM) mean frequency. A conversion of PPM into pulse amplitude modulation (PAM) in the receiver demodulation process permits a maximum information bandwidth to be transmitted at a given sampling rate. 7 refs.

51/7/20 (Item 8 from file: 8)
DIALOG(R)File 8:Ei Compendex(R)

01022399 E.I. Monthly No: EI8105044319 E.I. Yearly No: EI81086745
Title: ANALOG ENCODED CHIRP TRANSMISSION SYSTEM USING SURFACE ACOUSTIC WAVE FILTERS .
Author: Kowatsch, Max; Lafferl, Johann; Seifert, Franz J.; Stocker, Helmut R.
Corporate Source: Tech Univ, Vienna, Austria
Source: IEEE Transactions on Sonics and Ultransonics v SU-27 n 6 Nov 1980 p 355-359
Publication Year: 1980
CODEN: IESUAU ISSN: 0013-4902
Language: ENGLISH
Journal Announcement: 8105

Abstract: Linear frequency modulation is implemented in encoding analog signals. Using surface acoustic wave (SAW) dispersive delay lines, the transmission system described operates with overlapping chirps that are position- modulated according to the sampled data. Demodulation in the receiver requires pulse compression in a matched filter . The principle of analog encoding and decoding is explained. Experimental results are provided on the performance of a modem designed for handling baseband signals with a spectral range from 300 Hz to 400 kHz. SAW linear FM filters , operating at a center frequency of 60 MHz with a time-bandwidth product of 180, are employed. The system shows good amplitude and phase characteristics and a dynamic range greater than 40 dB. 9 refs.

51/7/21 (Item 1 from file: 94)
DIALOG(R)File 94:JICST-EPlus

01036375 JICST ACCESSION NUMBER: 90A0346595 FILE SEGMENT: JICST-E
Characteristics of land mobile earth stations for the EMSS project.
HAMAMOTO NAOKAZU (1); SUZUKI RYUTARO (1); IKEGAMI TETSUSHI (1); TAIRA SHIN'ICHI (1); ICHIYOSHI OSAMU (2); SATO NOBUYASU (3); TADA SHUN'ICHI (4)
(1) Tsushinsoken; (2) NEC Corp.; (3) Toshiba Corp., Komukai Works; (4) Kenwood Corp.

Tsushin Sogo Kenkyujo Kiho(Review of the Communications Research Laboratory), 1990, VOL.36,NO.Special Issue 10, PAGE.75-82, FIG.9, TBL.4, REF.5
JOURNAL NUMBER: F0415ABQ ISSN NO: 0914-9279
UNIVERSAL DECIMAL CLASSIFICATION: 621.396.73 621.396.946
LANGUAGE: Japanese COUNTRY OF PUBLICATION: Japan
DOCUMENT TYPE: Journal
ARTICLE TYPE: Original paper
MEDIA TYPE: Printed Publication

ABSTRACT: A feasibility study on land mobile satellite communication was carried out using three kinds of mobile earth stations. These earth stations were developed after considering the potential effect of such signal propagation characteristics as multi-pass fading and the blockage effect on future practical applications. The first earth station employs the ACSSB(Amplitude Companded SSB) technique, which can transmit voice signal with narrow bandwidth. The second earth station uses a digital signal processing technique applied to both QPSK modulation / demodulation and voice coding/decoding. The third earth station employs the spread spectrum(SS) system using a matched filter technique for de-spreading a receiving signal. This paper introduces the configurations of these earth stations and also reports on their performance. Additionally, an example of signal propagation characteristics in an urban area is presented. These are experimental results using the ETS-V satellite. (author abst.)

51/7/22 (Item 1 from file: 99)
DIALOG(R)File 99:Wilson Appl. Sci & Tech Abs

1203552 H.W. WILSON RECORD NUMBER: BAST94072215
Watkins-Johnson HF-1000 general-coverage receiver
Newkirk, David;
QST v. 78 (Dec. '94) p. 76-9
DOCUMENT TYPE: Product Evaluation ISSN: 0033-4812

ABSTRACT: The Watkins-Johnson HF-1000 general-coverage receiver is reviewed. It features a digital signal processor instead of analog circuitry to handle intermediate-frequency (IF) filtering , automatic gain control, and demodulation . The receiver covers 5 kHz to 30 MHz and can handle amplitude modulation (AM), synchronous AM, narrowband FM, continuous wave, upper sideband, lower sideband, and independent sideband. The receiver has a high-dynamic-range front end, a switchable radio-frequency (RF) preamplifier, a switchable RF attenuator, 58 IF bandwidth choices, a signal meter, a noise blanker, an IF notch filter , manual and automatic gain control, and squelch and scanning features. The HF-1000 is better suited to high-quality broadcast reception than demanding all-mode communication. The suggested retail price is $3,995.

51/7/23 (Item 1 from file: 108)
DIALOG(R)File 108:Aerospace Database

02098709 N93-12207
All-digital multicarrier demodulators for on-board processing satellites in mobile communication systems
Ph.D. Thesis
YIM, WAN HUNG

Surrey Univ., London (England).
CORPORATE CODE: S5540976
1991 230P.
LANGUAGE: English
COUNTRY OF ORIGIN: United Kingdom COUNTRY OF PUBLICATION: United Kingdom
DOCUMENT TYPE: THESIS
DOCUMENTS AVAILABLE FROM AIAA Technical Library
OTHER AVAILABILITY: Univ. Microfilms Order No. BRDX94966
JOURNAL ANNOUNCEMENT: STAR9302

Economical operation of future satellite systems for mobile communications can only be fulfilled by using dedicated on-board processing satellites, which would allow both cheap earth terminals and lower space segment costs. With on-board modems and codecs, the up-link and down-link can be optimized separately. An attractive scheme is to use frequency-division multiple access/single chanel per carrier (FDMA/SCPC) on the up-link and time division multiplexing (TDM) on the down-link. This scheme allows mobile terminals to transmit a narrow band, low power signal, resulting in smaller dishes and high power amplifiers (HPA's) with lower output power. On the up-link, there are hundreds to thousands of FDM channels to be demodulated on-board. The most promising approach is the use of all-digital multicarrier demodulators (MCD's), where analog and digital hardware are efficiently shared among channels, and digital signal processing (DSP) is used at an early stage to take advantage of very large scale integration (VLSI) implementation. A MCD consists of a channellizer for separation of frequency division multiplexing (FDM) channels, followed by individual modulators for each channel. Major research areas in MCD's are in multirate DSP, and the optimal estimation for synchronization, which form the basis of the thesis.

Complex signal theories are central to the development of structured approaches for the sampling and processing of bandpass signals, which are the foundations in both channellizer and demodulator design. In multirate DSP, polyphase theories replace many ad-hoc, tedious and error-prone design procedures. For example, a polyphase-matrix deep space network frequency and timing system (DFT) channellizer includes all efficient filter bank techniques as special cases. Also, a polyphase-lattice filter is derived, not only for sampling rate conversion, but also capable of sampling phase variation, which is required for symbol timing adjustment in all-digital demodulators. In modulation schemes, a systematic survey is reported, based on two expressions that includes all formats in linear and constant envelope modulation. In synchronization techniques, classifications according to the criterion of statistical optimization, the data dependecy, and the method of parameter extraction, reflect the inherent complexity and performance of numerous existing algorithms. The designs of two new algorithms are presented: a differential decision frequency error detector that is simple and fast; a dual-comb- filter frequency/timing error detector that is targeted at VLSI implementation. The real-time implementation of a complete 4 x 16 kb/s MCD for the T-SAT project is described in detail, which proved many of the structured design concepts developed in this thesis. The requirements of software tools for various levels of simulation in multirate DSP and communications are analyzed. This led to the implementation of a data-flow oriented simulation system, which was used in all research work in the thesis (Dissert. Abstr.)

51/7/24 (Item 2 from file: 108)
DIALOG(R)File 108:Aerospace Database

01472458 A84-26742

Algorithm implementation in a tactical communication brassboard processor

MYERS, M. H.; KORGEL, C. C. (TRW, Inc., Military Electronics Div., San Diego, CA)

IN: Digital Avionics Systems Conference, 5th, Seattle, WA, October 31-November 3, 1983, Proceedings (A84-26701 11-06). New York, Institute of Electrical and Electronics Engineers, 1983, p. 10.4.1-10.4.5.

1983

CONTRACT NO.: N00019-82-C-0330

LANGUAGE: English

COUNTRY OF ORIGIN: United States COUNTRY OF PUBLICATION: United States

DOCUMENT TYPE: CONFERENCE PAPER

DOCUMENTS AVAILABLE FROM AIAA Technical Library

JOURNAL ANNOUNCEMENT: IAA8411

This paper reports on the implementation of waveform processing algorithms in a high speed digital signal processor, utilizing newly developed VLSI component technology and containing architecture elements intended for the integration of avionics CNI terminals operating in the 2 MHz-2 GHz frequency band. The algorithms are executed by the combination of a set of reconfigurable waveform preprocessors controlled by a very high speed programmable signal processor. The algorithms process the waveforms and perform the basic terminal functions of JTIDS Class II DTDMA, GPS Navigation Satellite, HF-ECCM (HFIP), and clear voice AM or FM demodulation. Prioritized sharing of modular elements requiring scheduling of their resources is controlled by the signal processor. An important goal of this program is to verify that the combination of architecture and component technology selected would be a suitable candidate to replace many single function CNI avionics terminals carried in military aircraft. (Author)

SOURCE OF ABSTRACT/SUBFILE: AIAA

51/7/25 (Item 3 from file: 108)

DIALOG(R)File 108:Aerospace Database

00785120 A76-27639

Hybrid modem for 120 channels FDM telephony signal transmission

FURUYA, T.; SUGIYAMA, M.; MATSUO, Y.; KATAGIRI, Y.; TANAKA, H. (Nippon Electric Co., Ltd., Tokyo, Japan)

In: International Conference on Digital Satellite Communications, 3rd, Kyoto, Japan, November 11-13, 1975, Proceedings. (A76-27626 12-32) Tokyo, Kokusai Denshin Denwa Co., Ltd., 1975, p. 91-98. Research sponsored by the International Telecommunications Satellite Organization.

1975

LANGUAGE: English

COUNTRY OF ORIGIN: Japan COUNTRY OF PUBLICATION: Japan

DOCUMENT TYPE: CONFERENCE PAPER

JOURNAL ANNOUNCEMENT: IAA7612

The design of a hybrid modem using 2 pulse amplitude -phase modulation and the modem performance data are described. The hybrid modem can transmit 120 FDM telephony channels over a 2.5 MHz RF bandwidth. Coherence recovery and automatic level control are described in addition to the monitor circuit and signal processing and detection behavior. Data on signal degradation are tabulated. Weighted SNR 53 dB and 50 dB were obtained at CNR 16.9 dB and 13.9 dB respectively. The eye pattern seen at the output of the receive shaping filters and the recovered map seen at the output of the receive 8-bit A/D converters are reproduced. (R.D.V.)

SOURCE OF ABSTRACT/SUBFILE: AIAA

51/7/26   (Item 1 from file: 434)
DIALOG(R)File 434:SciSearch(R) Cited Ref Sci

14112745   Genuine Article#: RQ550   Number of References: 18
Title: ADAPTIVE CROSSTALK CANCELLATION AND LASER FREQUENCY DRIFT COMPENSATION IN DENSE WDM NETWORKS
Author(s): MINARDI MJ; INGRAM MA
Corporate Source: USAF,WRIGHT LAB,RADAR BRANCH,AARM/WRIGHT PATTERSON AFB//OH/45433; GEORGIA INST TECHNOL,SCH ELECT & COMP ENGN/ATLANTA//GA/30332
Journal: JOURNAL OF LIGHTWAVE TECHNOLOGY, 1995, V13, N8 (AUG), P1624-1635
ISSN: 0733-8724
Language: ENGLISH   Document Type: ARTICLE
Abstract: Two variations of the LMS algorithm are proposed to cancel received linear crosstalk in dense WDM networks. Analysis and simulations show that with the addition of a few photodetectors, channel spacing requirements can be reduced by over 50 percent. Simulations using a demultiplexer with Gaussian bandpass characteristics show that a 2.5 dB signal-to-crosstalk-plus-noise ratio can be increased to over 35 dB. The decision-directed algorithm will work with OOK data or any intensity modulation scheme which uses the absence of light as one symbol. The decision directed algorithm makes assumptions on the desired laser frequency so it can accommodate only limited laser drift. A second cancellation algorithm uses pilot tones added to each laser signal in order to cancel the crosstalk. It will work with analog or digital intensity- modulated data and will automatically configure itself to account for laser drift. Both algorithms are blind in that they do not require training sequences for initialization, Analysis shows that weights derived from pilot tones are nearly optimum for canceling crosstalk for the data, Simulations of both algorithms are presented. Finally, both algorithms are shown to be capable of canceling nonlinear beating terms.
?

```
File 237:Buyer's Guide to Micro Software(SOFT) 1993/Sep
        (c) 1993 ONLINE Inc.
File 256:SoftBase:Reviews,Companies&Prods. 95-1996/Jul
        (c)1996 Info.Sources Inc
File 278:Microcomput.Software Guide 1996/Aug
        (c) 1996 Reed Reference Publishing
File 751:Datapro Software Directory 1996/Jul
        (c) 1996 McGraw-Hill, Inc.
?ds

Set     Items   Description
S1       4534   VOICE OR AUDIO OR ANALOG?
S2         88   S1(2N)SIGNAL?
S3       8307   COMPRESS? OR REDUC? OR REDN?
S4         13   SUBCHANNEL? OR SUBCIRCUIT? OR SUBFACILIT? OR SUBLINE? OR S-
              UBPATH? OR SUBLINK? OR SUB()(CIRCUIT? OR FACILIT? OR LINE? OR
              PATH? OR LINK?)
S5      22921   TIME
S6          0   SINGLE(1W)(SIDEBAND? OR SIDE()BAND? ?)
S7          0   S2(4N)S3 AND S4 AND S5
S8          0   S7 AND S6
S9          0   (BANDWID? OR BAND()WIDTH)(4N)S3 AND S4
S10         0   (BANDWID? OR BAND()WIDTH)(4N)S3 AND SUB()CHANNEL?
S11         0   S2(4N)S3 AND S5 AND SUB()CHANNEL?
S12        10   BASESTATION? OR LANDSTATION? OR (BASE OR LAND)()STATION?
S13         0   S12 AND S1(4N)S3 AND S5
S14         0   S12 AND S1(4N)S3 AND S6
S15         0   S1(4N)S3 AND S4 AND S5
S16         0   S1(4N)S3 AND SUB()CHANNEL? AND S5
S17         3   S1(4N)S3 AND RECEIV?
S18         1   S17 AND S5
S19         0   S17 AND S6
S20      3139   DEMODULAT? OR DE()MODULAT? OR MODEM? OR S1(3N)DIGIT?
S21         1   S17 AND S20
S22         0   S18 AND (S19 OR S21)
S23      9344   PAGE OR RADIOPAG? OR PAGING OR BEEP? OR PAGE? ? OR RADIO? -
              ?(2N)(SUMMON? OR PRECONI?)
S24         0   S23 AND (S4 OR SUB()CHANNEL?)
S25         0   S24 AND (AMPLI? OR S5)
S26         0   S17 AND S6
S27         0   S21 AND 6
S28         0   S21 AND S6
S29       892   S20 AND (RECEIV? OR TERMINAL?)

S32         4   S29 AND FILTER?
S33         0   S32 AND (MODULAT? OR QAM)
S34         0   S33 AND AMPLI?
S35         0   S34 AND SIGNAL?
S36         0   S34 AND SIGNAL?(2N)PROCESS?
?
t18/7/1
>>> PAUSE ended.

 18/7/1      (Item 1 from file: 256)
DIALOG(R)File 256:SoftBase:Reviews,Companies&Prods.
(c)1996 Info.Sources Inc. All rts. reserv.

00062244          DOCUMENT TYPE:  Review

PRODUCT NAMES:  David OS-9  (499196); David OS-9000  (499196)
```

TITLE: Real- Time OS, Services Drive TV Decoders
AUTHOR: Miller, Eric
SOURCE: Electronic Design, v42 n4 pES24(3) Feb 21, 1994 0013-4872

RECORD TYPE: Review
REVIEW TYPE: Product Analysis
GRADE: Product Analysis, No Rating

Set-top box design for next generation, interactive digital TV services requires transformation from analog technology to real- time computing with duplexed communications and high-bandwidth data transfer. Real- time OSs and services must manage and synchronize compressed video and audio . The computer must execute an application program containing various simultaneous tasks, such as providing a user interface and program selection methods. Dedicated set-top boxes will receive incoming signals and decode compressed bit streams, including MPEG streams. David is a decoder box and software package that includes OS-9 and OS-9000 OSs and system service modules for digital video applications. Software is based on a real- time kernel that communicates with real- time audio/video and an I/O manager, which runs system service modules.

REVISION DATE: 940823
?t21/7/1

21/7/1 (Item 1 from file: 256)
DIALOG(R)File 256:SoftBase:Reviews,Companies&Prods.

00078323 DOCUMENT TYPE: Review

PRODUCT NAMES: Verilog Compiled Simulator (VCS) (490105); Verilog-XL (476871); Motion Picture Experts Group (MPEG) (832146)

TITLE: General Instrument Designs A Set Top Box Chipset
AUTHOR: Balthazor, Brian Lodman, Mike Pollmann, Steve Price, Rick...
SOURCE: Integrated System Design, p32(7) Mar 1995 1067-9804

RECORD TYPE: Review
REVIEW TYPE: Product Analysis
GRADE: Product Analysis, No Rating

General Instrument designed a five chip set DigiChip II Decoder implementation for satellite and broadcast terrestrial applications. In the system, an encoder digitizes and compresses video and audio , and combines it with access control information. It is then error correction encoded, and sent to the satellite. The satellite stream is subsequently received by a satellite dish, broadcast station, or cable headend. The receiver demodulates , and does error correction and decompression. Five ASICs in the application include the forward error correction processor. The access control processor accepts the MPEG-2 stream, and authenticates requested services. This processor then routes the streams to the graphics transport processor and video decompression processor. The graphics processor interprets audio MPEG-2 transport layers, and routes audio data to the audio decompression processor. General Instrument used C, Verilog, and VCS to design the system.

REVISION DATE: 951030
?

```
File  35:Dissertation Abstracts Online 1861-1996/Sep
         (c) 1996 UMI
File 202:Information Science Abs. 1966-1996/Jul
         (c) 1996 IFI/Plenum Data Corp.
File 233:Microcomputer Abstracts(TM) 81-1996/Aug
         (c) 1996 Information Today,Inc
?ds

Set     Items   Description
S1      45314   VOICE OR AUDIO OR ANALOG?
S2        953   S1(2N)SIGNAL?
S3     118129   COMPRESS? OR REDUC? OR REDN?
S4        465   SUBCHANNEL? OR SUBCIRCUIT? OR SUBFACILIT? OR SUBLINE? OR S-
               UBPATH? OR SUBLINK? OR SUB()(CIRCUIT? OR FACILIT? OR LINE? OR
               PATH? OR LINK?)
S5     195316   TIME
S6         49   SINGLE(1W)(SIDEBAND? OR SIDE()BAND? ?)
S7          0   S2(4N)S3 AND S4 AND S5
S8          0   S7 AND S6
S9          0   (BANDWID? OR BAND()WIDTH)(4N)S3 AND S4
S10         0   (BANDWID? OR BAND()WIDTH)(4N)S3 AND SUB()CHANNEL?
S11         0   S2(4N)S3 AND S5 AND SUB()CHANNEL?
S12       174   BASESTATION? OR LANDSTATION? OR (BASE OR LAND)()STATION?
S13         0   S12 AND S1(4N)S3 AND S5
S14         0   S12 AND S1(4N)S3 AND S6
S15         0   S1(4N)S3 AND S4 AND S5
S16         0   S1(4N)S3 AND SUB()CHANNEL? AND S5
S17        33   S1(4N)S3 AND RECEIV?
S18         8   S17 AND S5
S19         1   S17 AND S6
S20     11826   DEMODULAT? OR DE()MODULAT? OR MODEM? OR S1(3N)DIGIT?
S21        16   S17 AND S20
S22         4   S18 AND (S19 OR S21)
?
t22/7/all
>>> PAUSE ended.
```

22/7/1   (Item 1 from file: 202)
DIALOG(R)File 202:Information Science Abs.

00178046 9308046
ISA Document Number in Printed Publication: 9308174
Signal computing.
Document Type: Journal Article
Author (Affiliation): Anderson, E.C.; Shepard, S.; Sohn, P.
Journal: Byte
Publication Language(s): English
Source: Vol. 17 Issue 12 p. 155-164 Nov 1992

Digital signal processing (DSP) enables computers to transform real-world signals in either audio, video, or electromagnetic form into a practicable form. This paper defines the types of signals received and processed by the DSP (composite, digital, analog, discrete- time and -amplitude) and describes the processes involved: A/D conversion, Fourier transformation, Nyquist theorem, signal reconstruction, digital filtering, data compression, sample-rate converters, and adaptive filtering.

22/7/2   (Item 1 from file: 233)

DIALOG(R)File 233:Microcomputer Abstracts(TM)

0406408 95OV12-006
Audio and video online
Online & CD-ROM Review , December 1, 1995 , v19 n6 p347-349, 3 Page(s)
ISSN: 0309-314X
Focuses on recent products and developments for accessing digital audio and video, noting that information providers can incorporate sound and vision into online offerings without users needing high-speed connections such as ISDN. Covers the RealAudio Player, demonstrated by Microsoft, which allows PCs with a 14.4Kbps modem to play a compressed audio file as it is downloaded from an Internet server, and it is available free over the Internet. Explains that Streamworks ($29) from Xing Technology of Arroyo Grande, CA, compresses one second of speech into 1K of data, and it allows computer users to hear what is being broadcast at the same time to conventional radio receivers. Notes that the NBC Multimedia Player is software that allows audio and video to be played in real time over the WWW using a 14.4Kbps modem, and the Internet Phone from Vocaltec offers real-time telephone connection over the Internet. Includes one sidebar, three screen displays, one photo, and a list of vendors. (jo)

22/7/3 (Item 2 from file: 233)
DIALOG(R)File 233:Microcomputer Abstracts(TM)

0390893 95IE07-001
Net as phone: kiss long-distance charges good-bye
Savetz, Kevin
Internet World , July 1, 1995 , v6 n7 p67-69, 3 Page(s) ISSN: 1064-3923
Discusses using the Internet to hold long distance phone conversations. Says using new technologies such as audioconferencing allows users to avoid expensive calling charges when they want to talk to somebody on the other side of the world. Because speech takes more bandwidth than a normal analog modem can provide, the speech is compressed for transmission and then uncompressed when the other computer receives it. The software is also plagued by gaps in transmission time and many programs allow only one person at a time to speak, similar to a CB radio. The compression also makes the audio lose some quality, although the quality loss can be reduced by listening through headphones. Includes a sidebar. (eqb)

22/7/4 (Item 3 from file: 233)
DIALOG(R)File 233:Microcomputer Abstracts(TM)

0375619 95PK02-102
Telecom, digital limits begin to blur with 'phone calls' across Internet
Patrizio, Andy
PC WEEK , February 13, 1995 , v12 n6 p1, 114, 2 Page(s) ISSN: 0740-1604
Company Name: VocalTec
Product Name: Internet Phone
Indicates that the release of Internet Phone ($49) from VocalTec of Northvale, NJ, will provide a new audio-compression technology that allows users to make long-distance ''phone calls'' across the Internet for the cost of a local call. Explains that this communications package transports voice packets in real time, resulting in users thus able to have voice

conversations over the Incernet anywhere in the worlu. Notes that Internet Phone is less bandwidth-intensive than downloading files due to digital aud being more of a burst than a constant stream. Reports that In Phone works with any Winsock-compliant application over connectio ranging from a 14.4Kbps SLIP line, to a dedicated T-3 line. Howev says that voice calls made over the Internet do not have the qualit of those over phone lines, there may be security issues, and Internet Phone cannot receive voice packets from outside a firewall. Includes one diagram and one screen display. (jo)
?

Set	Items	Description
S1	45314	VOICE OR AUDIO OR ANALOG?
S3	118129	COMPRESS? OR REDUC? OR REDN?
S17	33	S1(4N)S3 AND RECEIV?
S6	49	SINGLE(1W)(SIDEBAND? OR SIDE()BAND? ?)
S19	1	S17 AND S6

t19/7/1

19/7/1 (Item 1 from file: 35)
DIALOG(R)File 35:Dissertation Abstracts Online

821810 ORDER NO: AAD83-20746
THE SATELLITE APPLICATIONS OF AMPLITUDE COMPANDED SINGLE SIDEBAND MODULATION

Author: MEHRING, JAMES WARREN
Degree: PH.D.
Year: 1983
Corporate Source/Institution: STANFORD UNIVERSITY (0212)
Source: VOLUME 44/05-B OF DISSERTATION ABSTRACTS INTERNATIONAL. PAGE 1544. 312 PAGES

Amplitude Companded Single Sideband or ACSB is a voice processing technique that was originally developed at Stanford University for mobile radio. Previous studies on ACSB for mobile radio have shown a substantial subjective voice improvement for good receive carrier-to-noise ratios. These results for mobile radio indicate that the power and noise figures available on satellite links are sufficient to use ACSB modulation. Single sideband modulation is much more spectrally efficient than currently used modulation techniques and increased channel availability was projected. This study demonstrated that ACSB can be used on satellite links and that its use makes possible an order of magnitude increase in telephone channel availability on existing links.

ACSB processing consists of an initial stage of ratio-2 companding, a stage of pre-emphasis, pilot injection and a final stage of ratio-2 companding. The compandors logarithmically reduce the dynamic range of the input signal. Voice compression results in the lower amplitude portions of speech being transmitted with more power. This results in signal-to-noise ratio improvement for the low amplitude sounds thereby providing significant subjective voice quality improvement. During pauses between syllables, words, or speech passages, the only input to the receive ACSB processing equipment is channel noise. The expansion process of the receive compandors suppresses the noise and provides subjective improvement by increasing the contrast between voice and background noise. Single sideband also provides power improvement from the fact that the power in the sideband is directly proportional to the input voice power. This provides sharing between channels on multichannel ACSB systems. Analysis and experimental measurements conservatively estimate this improvement at 7 dB.

Computer simulation was used to model the ACSB processing to optimize the processing sequence for satellite use and predict performance for voice and voiceband data signals. The sensitivity of ACSB to intermodulation and frequency error was studied. The ACSB circuits were constructed and their performance was demonstrated at audio, 27 MHz, 57 MHz and through a transponder on WESTAR IV. The testing indicated a 12 dB pilot

was optimum for satellite applications and that the minimum zero margin receive peak syllable power-to-noise ratio was 19 dB.

?

```
S4          465  SUBCHANNEL? OR SUBCIRCUIT? OR SUBFACILIT? OR SUBLINE? OR S-
                 UBPATH? OR SUBLINK? OR SUB()(CIRCUIT? OR FACILIT? OR LINE? OR
                 PATH? OR LINK?)
S5       195316  TIME

S23       16087  PAGE OR RADIOPAG? OR PAGING OR BEEP? OR PAGE? ? OR RADIO? -
                 ?(2N)(SUMMON? OR PRECONI?)
S24           7  S23 AND (S4 OR SUB()CHANNEL?)
S25           2  S24 AND (AMPLI? OR S5)
?
?t25/6/all

 25/6/1     (Item 1 from file: 35)
01490694  ORDER NO: AADAA-INN05794
TRIE METHODS FOR TEXT AND SPATIAL DATA ON SECONDARY STORAGE


 25/6/2     (Item 2 from file: 35)
01096778  ORDER NO: AAD90-08967
MECHANISMS OF MULTIDRUG RESISTANCE: DIFFERENCES BETWEEN NATURAL AND
ACQUIRED ADRIAMYCIN-RESISTANT HUMAN COLON CARCINOMA CELLS (COLON CARCINOMA
CELLS)
?
t25/7/1

 25/7/1     (Item 1 from file: 35)
DIALOG(R)File  35:Dissertation Abstracts Online
(c) 1996 UMI. All rts. reserv.

01490694  ORDER NO: AADAA-INN05794
TRIE METHODS FOR TEXT AND SPATIAL DATA ON SECONDARY STORAGE
  Author:  SHANG, HEPING
  Degree:  PH.D.
  Year:    1995
  Corporate Source/Institution:  MCGILL UNIVERSITY (CANADA) (0781)
  Adviser: T. H. MERRETT
  Source:  VOLUME 57/03-B OF DISSERTATION ABSTRACTS INTERNATIONAL.
           PAGE 1921.  142 PAGES
  ISBN:           0-612-05794-1
```

This thesis presents three trie organizations for various binary tries. The new trie structures have two distinctive features: (1) they store no pointers and require two bits per node in the worst case, and (2) they partition tries into pages and are suitable for secondary storage. We apply trie structures to indexing, storing and querying both text and spatial data on secondary storage. We are interested in practical problems such as storage compactness, I/O efficiency, and large trie construction.

We use our tries to index and search arbitrary substrings of a text. For an index of 100 million keys, our trie is 10%-25% smaller than the best known method. This difference is important since the index size is crucial for trie methods. We provide methods for dynamic tries and allow texts to be changed. We also use our tries to compress and approximately search large dictionaries. Our algorithm can find strings with k mismatches in sublinear time. To our knowledge, no other published sublinear algorithm is known for this problem.

Besides, we use our tries to store and query spatial data such as maps. A trie structure is proposed to permit querying and retrieving spatial data at arbitrary levels of resolution, without reading from secondary storage any more data than is needed for the specified resolution. The trie structure also compresses spatial data substantially.

The performance results on map data have confirmed our expectations: the querying cost is linear in the amount of data needed and independent of the data size in practice. We give algorithms for a set of sample queries including geometrical selection, geometrical join and the nearest neighbour. We also show how to control query cost by specifying an acceptable resolution.
?

Set	Items	Description
S1	45314	VOICE OR AUDIO OR ANALOG?
S3	118129	COMPRESS? OR REDUC? OR REDN?
S6	49	SINGLE(1W)(SIDEBAND? OR SIDE()BAND? ?)
S17	33	S1(4N)S3 AND RECEIV?
S20	11826	DEMODULAT? OR DE()MODULAT? OR MODEM? OR S1(3N)DIGIT?
S21	16	S17 AND S20
S27	0	S21 AND 6

?

S28 0 S21 AND S6

?

```
S20      11826   DEMODULAT? OR DE()MODULAT? OR MODEM? OR S1(3N)DIGIT?
S29       1825   S20 AND (RECEIV? OR TERMINAL?)
S32         84   S29 AND FILTER?
S33         38   S32 AND (MODULAT? OR QAM)
S34         14   S33 AND AMPLI?
S36          5   S34 AND SIGNAL?(2N)PROCESS?
?
t36/7/all
>>> PAUSE ended.
```

36/7/1 (Item 1 from file: 35)
DIALOG(R)File 35:Dissertation Abstracts Online

01419010 ORDER NO: AADAA-I9518997
A 100 MHZ, 5-MBAUD QAM RECEIVER FOR DIGITAL TELEVISION APPLICATIONS (CATV, HDTV)
Author: JOSHI, ROBINDRA B.
Degree: PH.D.
Year: 1995
Corporate Source/Institution: UNIVERSITY OF CALIFORNIA, LOS ANGELES (0031)
Chair: HENRY SAMUELI
Source: VOLUME 56/02-B OF DISSERTATION ABSTRACTS INTERNATIONAL. PAGE 1004. 170 PAGES

Next generation digital television systems, such as proposed Cable Television (CATV) and High Definition Television (HDTV) services, will rely on transceivers to deliver data at rates of up to 40 Mb/s. Quadrature amplitude modulation (QAM) schemes, successfully employed in high speed modems and digital radio systems, represent a promising transmission format for CATV and HDTV systems. QAM is a passband transmission scheme noted for its high bandwidth efficiency and is therefore an attractive modulation scheme for CATV and HDTV, both of which must operate under limited bandwidth budgets. Proposed specifications for QAM digital television schemes are 16- QAM at a 20 Mb/s data rate for HDTV and 64- QAM at a 30 Mb/s data rate for CATV. The HDTV and CATV system broadcasts must be compatible with the existing 6 MHz NTSC channel bandwidths and thus 5 MBaud is the maximum practical symbol rate which can be transmitted in the existing channel slots.

This dissertation presents results from the investigation of the architecture design and VLSI implementation of a 5-Mbaud QAM receiver chip designed to accommodate digital television applications. The QAM receiver chip contains all of the signal , processing blocks required to implement a complete all-digital QAM receiver including a quadrature demodulator , an adaptive feed-forward equalizer (FFE), an adaptive decision-feedback equalizer (DFE), a sampling phase error detector, and a numerically controlled oscillator (NCO). The FFE and DFE are both 20 tap filters and use an LMS coefficient updating algorithm. The FFE can be used in either a T-spaced or T/2-spaced mode. The projected maximum clock frequency of the receiver chip is 100 MHz for a symbol rate of 5 Mbaud. The receiver accommodates modulation formats as complex as 256- QAM which corresponds to a peak throughput rate of 40 Mbits/s. The receiver was created as a two piece chipset and was fabricated in 1.0-micron CMOS.

36/7/2 (Item 2 from file: 35)
DIALOG(R)File 35:Disseftation Abstracts Online

01204929 ORDER NO: AAD92-06133
HYBRID ACOUSTO-OPTIC AND DIGITAL EQUALIZATION OF MICROWAVE DIGITAL RADIOS
Author: ANDERSON, CHRISTOPHER SCOTT
Degree: PH.D.
Year: 1991
Corporate Source/Institution: NORTH CAROLINA STATE UNIVERSITY AT RALEIGH (0155)
Director: ANTHONY VANDERLUGT
Source: VOLUME 52/09-B OF DISSERTATION ABSTRACTS INTERNATIONAL. PAGE 4882. 270 PAGES

Digital radio transmission systems use complex modulation schemes that require powerful signal - processing techniques to correct channel distortions and to minimize bit error rates. In this dissertation, a class of combined analog and digital processors is developed that minimizes the mean square error of the radio receiver . The analog filters are implemented using acousto-optic processing since rapidly adaptable, inverse channel filters can be produced. A specific architecture and algorithm is identified and a prototype processor is designed, constructed, and tested. Laboratory measurements verify the ability of the processor to track and correct time-varying channels. These measurements also allow accurate modeling of the acousto-optic processor in a microwave digital radio simulation that includes quadrature amplitude modulation , channel distortion, hybrid equalization, and mean square error timing recovery. Computer analysis shows that hybrid equalization allows a four-fold increase in the data capacity of radio, relative to all digital equalization, while maintaining or decreasing sensitivity to channel distortion. This additional capacity will increase the number of voice-band telecommunication signals that can be supported by a single radio.

36/7/3 (Item 3 from file: 35)
DIALOG(R)File 35:Dissertation Abstracts Online

01190876 ORDER NO: AADDX-93002
AN INVESTIGATION INTO THE EFFECTS OF USING LIMITED PRECISION INTEGER ARITHMETIC IN DIGITAL MODEMS
Author: HALE, R. G.
Degree: PH.D.
Year: 1990
Corporate Source/Institution: UNIVERSITY OF TECHNOLOGY, LOUGHBOROUGH (UNITED KINGDOM) (5027)
Source: VOLUME 52/04-B OF DISSERTATION ABSTRACTS INTERNATIONAL. PAGE 2210. 324 PAGES

Available from UMI in association with The British Library.

The main aim of this thesis is to study the effects of using a reduced level of arithmetical precision (as found in a 16-bit microprocessor) whilst running various algorithms in the detection stages of a digital modem . The reason for using a lower precision is to see if these algorithms will run on a limited precision device, such as a Texas Instruments TMS320C25 digital signal processor , in real time.

In the study, data is transmitted over voice-band channels, such as telephone circuits and H.F. radio links, where the main imparements are

intersymbol interference and additive noise. The characteristics of these channels is briefly studied.

The thesis includes a study of quadrature amplitude modulated (QAM) signals transmitted over voice-band channels when the transmission path has, both, time invariant characteristics and when it introduces Rayleigh fading into the transmitted signal. Based on this study, an equivalent baseband model of the QAM system is derived, which is used in extensive computer simulation tests.

Two near maximum likelihood detectors are studied, one of which is a modified version of the pseudobinary reduced state Viterbi algorithm and is easily implemented in real time. The linear feedforward channel estimator is also studied, along with an efficient root finding algorithm that adaptively adjusts a pre- filter placed ahead of the detector such that the combined impulse response of the channel and filter is minimum phase. To show the effects of using integer arithmetic, all of these algorithms are simulated using integer variables.

Results of computer simulation tests are presented, showing the performance of the above algorithms whilst they are running independently from each other and when they are combined to form a complete modem receiver . High precision numerical values, calculated using the NAG mathematical library, are used to demonstrate the accuracy of the root finding algorithm when it is running with integer arithmetic. Results of running the root finding algorithm on a TMS320C25 development board are also given. These results are also compared with those found using the NAG library.

36/7/4 (Item 4 from file: 35)
DIALOG(R)File 35:Dissertation Abstracts Online

0971289 ORDER NO: NOT AVAILABLE FROM UNIVERSITY MICROFILMS INT'L.
ANALOG OPTICAL PROCESSING OF RADIO FREQUENCY SIGNALS
Author: SULLIVAN, DANIEL PAUL
Degree: PH.D
Year: 1987
Corporate Source/Institution: UNIVERSITY OF SOUTHERN CALIFORNIA (0208)
Source: VOLUME 48/09-B OF DISSERTATION ABSTRACTS INTERNATIONAL. PAGE 2747.

This paper contains an approach to the implementation of optical processing in complex electronic systems intended to receive and operate on multiple radio frequency (or microwave) signals. The goal is to exploit the rapidly expanding exploration of linear and nonlinear optics to realize transponders and receiving systems for satellites and other platforms. The inputs are assumed to be microwave. The outputs are assumed to be microwave or electronic (digital). The focus of the effort is in the architecture for the electronic functions, that allow optical component synthesis. These elements accomplish the following signal processing operations: amplitude modulation and demodulation ; phase modulation and demodulation ; frequency modulation and demodulation ; phase locked loop signal tracking; carrier element mixing (frequency shifting); signal filtering ; and signal matched filtering .

To accomplish the above operations on a theoretical basis, hypothetical devices are assumed for spatial light modulation of the RF signals onto an optical beam, and for operating on one light beam complex envelope with another light beam complex envelope in four different forms. Candidate optical component realizations of these hypothetical devices are presented utilizing acousto-optic cells, photorefractive crystals, and

crystals with third order nonlinearities, including an original approach to an ultra-linear acousto-optic modulator. Current techniques in optics employed include Fourier plane filtering, holography, and four wave mixing.

Total optical implementations are developed for a frequency division multiplexed and frequency modulated microwave carrier, a tracking phase locked loop, and a radar pulse receiving system. The approach taken is to create optical analogs to the electronic system implementations which will lead to extensive growth in the replacement of electronics with optoelectronics, and eventually integrated optics. The results represent an existence proof of candidate architectures which will allow complete optical processing synthesis of communication and radar systems. (Copies available exclusively from Micrographics Department, Doheny Library, USC, Los Angeles, CA 90089-0182.)

36/7/5 (Item 1 from file: 202)
DIALOG(R)File 202:Information Science Abs.

00109391 8610689
ISA Document Number in Printed Publication: 8610689
Converting and decoding receiver for digital data recorded in analog form on magnetic tape (Patent).
Document Type: Patent
Author (Affiliation): Kammayer, K.-D.; Rungeler, A.
Patent Assignee(s): Robert Bosch, GMBH, DE
Patent Number(s): US 4646173
Publication Language(s): English
Source: Feb 24, 1987

Apparatus for receiving data transmitted or recorded synchronously in the form of a multistep-coded modulated wave signal modulated at least in part in multistep phase or frequency modulation, which data are subject to timing jitter and to sudden and short-interval, changes of amplitude after transmission or recording and before reception, said apparatus comprising an analog -to- digital converter, a quadrature pair of digital bandpass filters connected so that both said filters are supplied with output signals from said analog -to- digital converter, a decision circuit for selecting a complex data signal from a set of predetermined complex reference data signals which most nearly approximates a complex data signal constituted by contemporary outputs of said filters during intervals of predetermined length, said decision circuit being connected to the outputs of said filters by switching means defining said intervals and further comprising: means for clocking said analog -to- digital converter at a controllable rate in a frequency range in the neighborhood of a predetermined rate, determined from input signal characteristics expected from transmission or recording standards, in response to a timing control signal; means for deriving said timing control signal by comparison of input and output signals of said decision circuit and for applying said frequency control signal to said clocking means, and means provided in said filters for varying the propagation time therethrough of signals supplied by said analog -to- digital converter, and for accomplishing said variation in response to said timing control signal, said filters being constituted for providing stepwise variable propagation time therethrough without substantial change of filter bandwidth and being connected to said timing control signal deriving means for propagation time control in response to said control signal

US005689440A

United States Patent [19]

Leitch et al.

[11] Patent Number: 5,689,440

[45] Date of Patent: Nov. 18, 1997

[54] **VOICE COMPRESSION METHOD AND APPARATUS IN A COMMUNICATION SYSTEM**

[75] Inventors: **Clifford Dana Leitch**, Coral Springs; **Robert John Schwendeman**, Pompano Beach; **Kazimierz Siwiak**; **William Joseph Kuznicki**, both of Coral Springs; **Sunil Satyamurti**, Delray Beach, all of Fla.

[73] Assignee: **Motorola, Inc.**, Schaumburg, Ill.

[21] Appl. No.: **764,656**

[22] Filed: **Dec. 11, 1996**

Related U.S. Application Data

[63] Continuation of Ser. No. 395,747, Feb. 28, 1995, abandoned.

[51] **Int. Cl.**[6] **H04B 7/00**

[52] **U.S. Cl.** .... **364/514 R**; 370/109; 455/54.1; 455/70; 381/29; 381/34

[58] **Field of Search** .... 381/29, 30, 35, 381/36, 37, 34; 364/514 R; 395/2.14, 2.12, 2.21, 2.24, 2.29; 340/825.44; 455/54.1, 70, 72, 109, 47, 38.1

[56] **References Cited**

U.S. PATENT DOCUMENTS

3,793,589	2/1974	Puckette	325/137
4,134,069	1/1979	Shiki .	
4,586,174	4/1986	Wong	370/69.1
4,839,923	6/1989	Kotzin	381/31
4,875,038	10/1989	Siwiak et al.	340/825.44
4,882,579	11/1989	Siwiak	340/825.44
4,955,083	9/1990	Phillips et al. .	
5,068,898	11/1991	Dejmek et al.	381/29
5,121,391	6/1992	Paneth et al. .	
5,175,769	12/1992	Hejna, Jr. et al.	340/825.44
5,216,744	6/1993	Alleyne et al.	381/29
5,239,306	8/1993	Siwiak et al.	340/825.44
5,282,205	1/1994	Kuznicki	370/94.1
5,387,981	2/1995	Oden et al.	358/400
5,533,062	7/1996	Liberti, Jr. et al.	375/334
5,535,215	7/1996	Hieatt, III	370/95.1

OTHER PUBLICATIONS

Verhelst and Roelands, *An Overlap–Add Technique Based On Waveform Similarity (WSOLA) For High Qality Time–Scale Modification Of Speech*, IEEE 1993, pp. II–554–II–557.

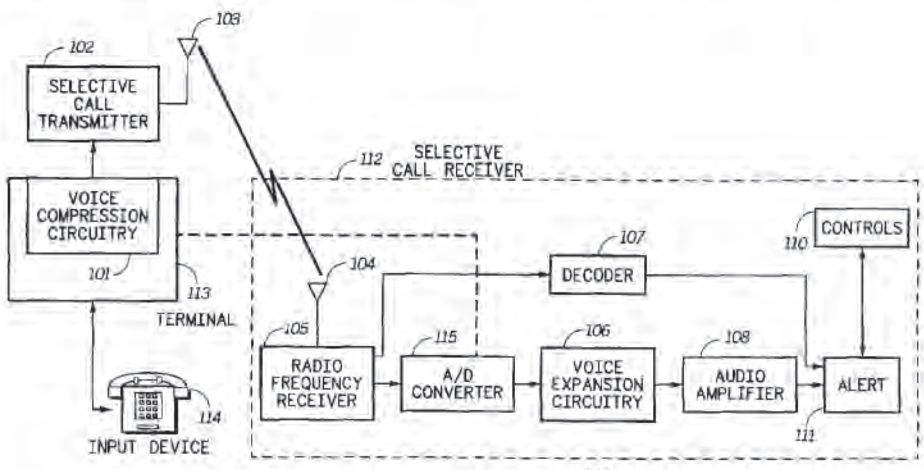
Oppenheim and Schafer, *Changing the Sampling Rate Using Discrete–Time Processing*, Discrete–Time Signal Processing, 1989, Ch. 3.6, pp. 101–112.

Primary Examiner—Ellis B. Ramirez
Assistant Examiner—Demetra R. Smith
Attorney, Agent, or Firm—James A. Lamb

[57] **ABSTRACT**

The present invention comprises a method for compressing a plurality of voice signals within a voice communication resource (see FIG. 6) having a given bandwidth within a voice communication system (**100**). The method comprises the steps of subchanneling the voice communication resource into a plurality of subchannels (**441**, **442**, **443**), placing a pair of the plurality of voice signals (**401**, **402**) on a subchannel (**441**); modulating the pair of the plurality of voice signals (**401**, **402**) about a pilot signal (**581**) within the subchannel (**441**) using single sideband modulation; and compressing the time of each of the voice signals (**401**, **402**) within the plurality of subchannels (**441**, **442**, **443**), wherein these step provide a compressed voice signal.

20 Claims, 14 Drawing Sheets



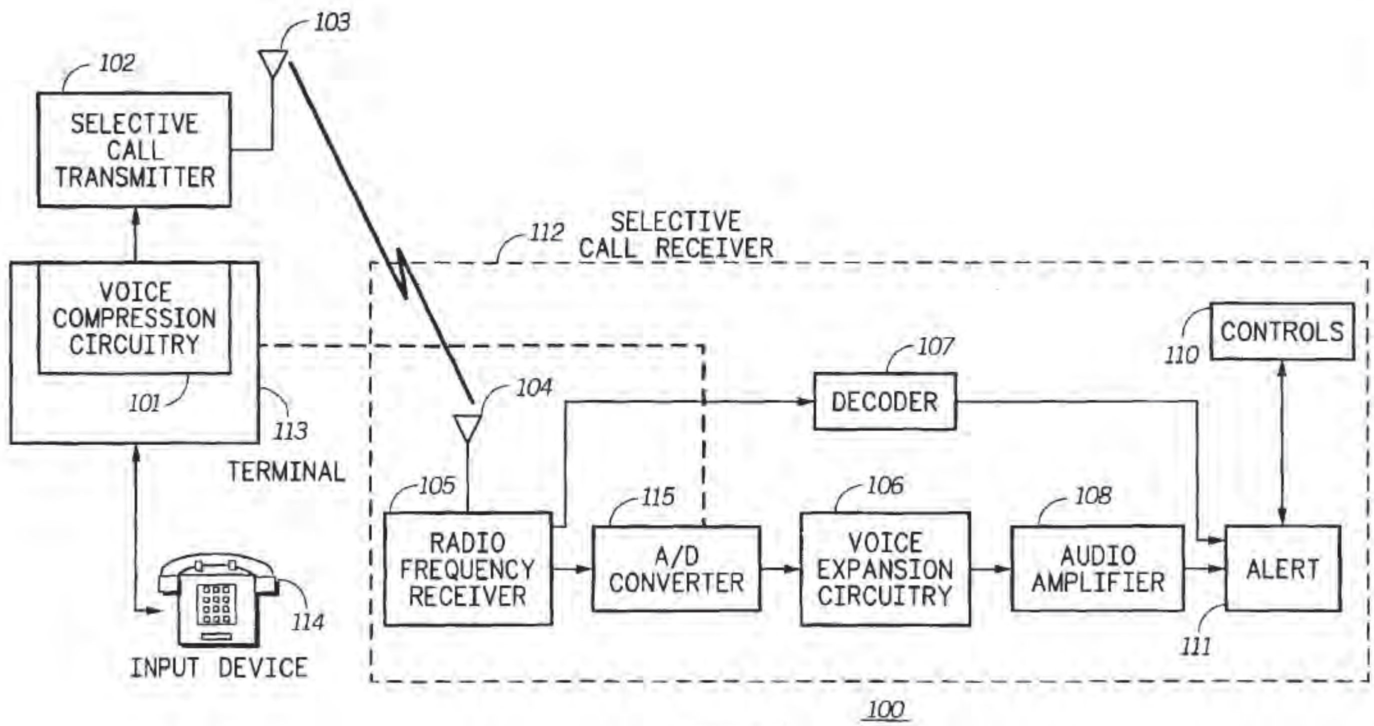

102
SELECTIVE CALL TRANSMITTER
103
VOICE COMPRESSION CIRCUITRY
101
113
TERMINAL
112
SELECTIVE CALL RECEIVER
104
105
RADIO FREQUENCY RECEIVER
115
A/D CONVERTER
107
DECODER
106
VOICE EXPANSION CIRCUITRY
108
AUDIO AMPLIFIER
110
CONTROLS
ALERT
111
114
INPUT DEVICE
100

FIG.1

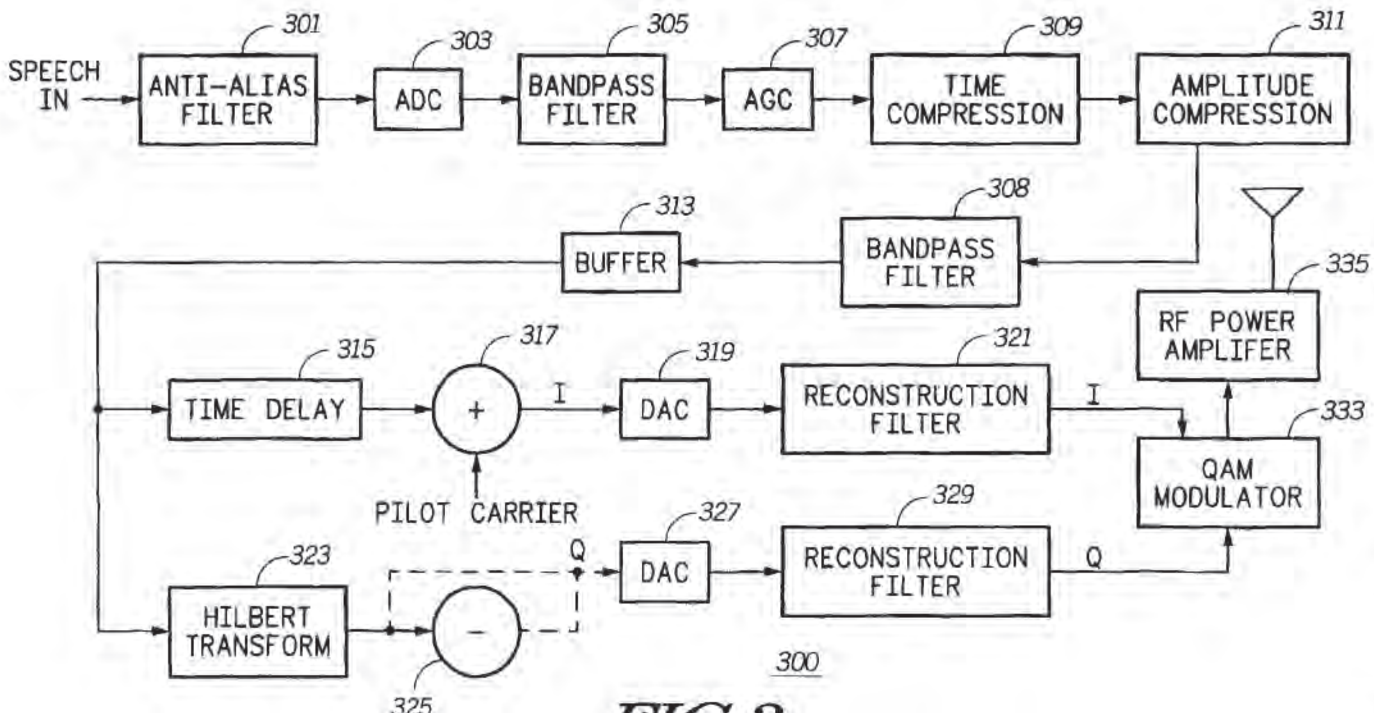

SPEECH IN
ANTI-ALIAS FILTER
301
ADC
303
BANDPASS FILTER
305
AGC
307
TIME COMPRESSION
309
AMPLITUDE COMPRESSION
311
BANDPASS FILTER
308
BUFFER
313
TIME DELAY
315
+
317
I
DAC
319
RECONSTRUCTION FILTER
321
I
PILOT CARRIER
RF POWER AMPLIFER
335
QAM MODULATOR
333
HILBERT TRANSFORM
323
−
325
Q
DAC
327
RECONSTRUCTION FILTER
329
Q
300

FIG.3

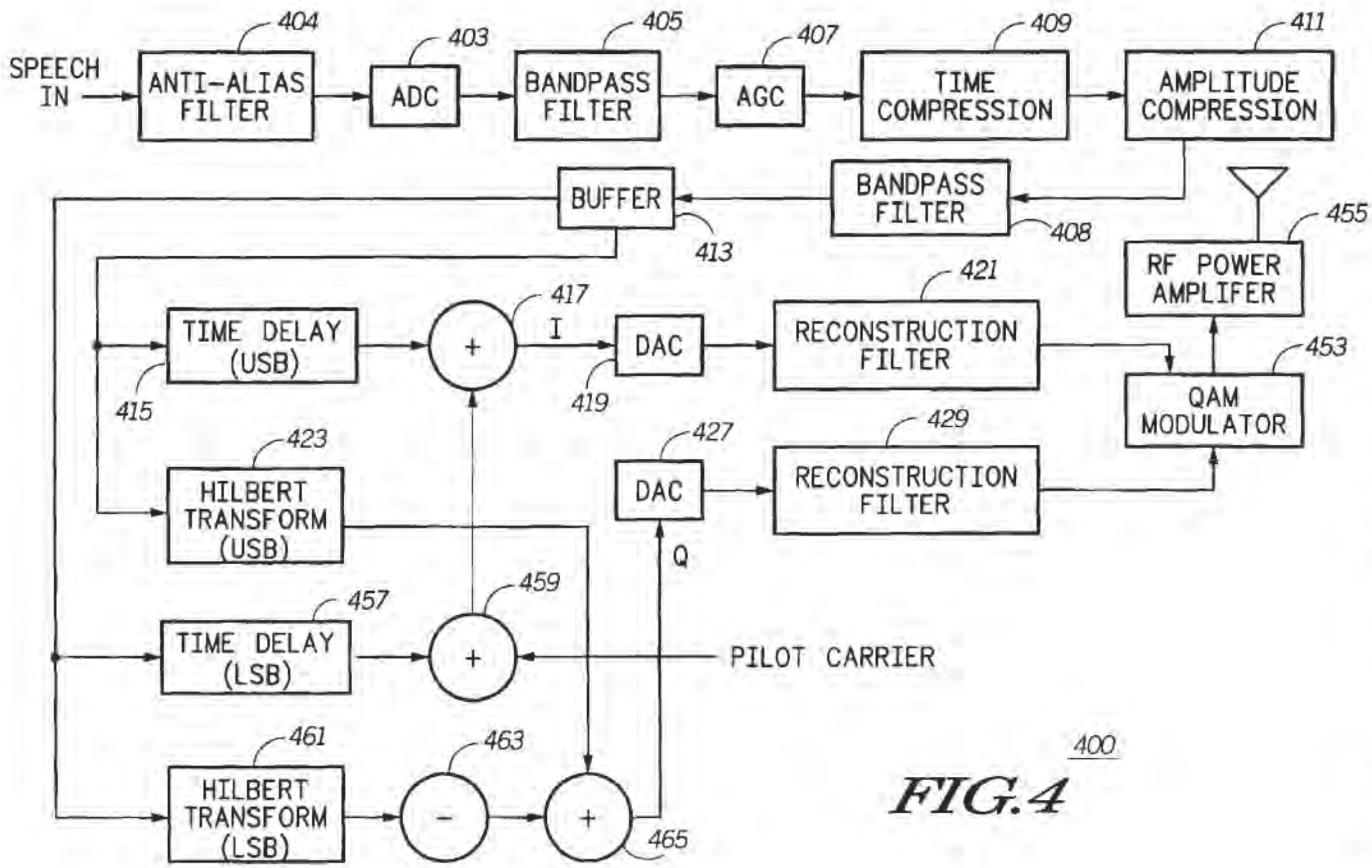

SPEECH IN
ANTI-ALIAS FILTER
404
ADC
403
BANDPASS FILTER
405
AGC
407
TIME COMPRESSION
409
AMPLITUDE COMPRESSION
411
BUFFER
413
BANDPASS FILTER
408
RF POWER AMPLIFER
455
TIME DELAY (USB)
415
+
417
I
DAC
419
RECONSTRUCTION FILTER
421
QAM MODULATOR
453
HILBERT TRANSFORM (USB)
423
DAC
427
Q
RECONSTRUCTION FILTER
429
TIME DELAY (LSB)
457
+
459
PILOT CARRIER
HILBERT TRANSFORM (LSB)
461
-
463
+
465
400

FIG.4

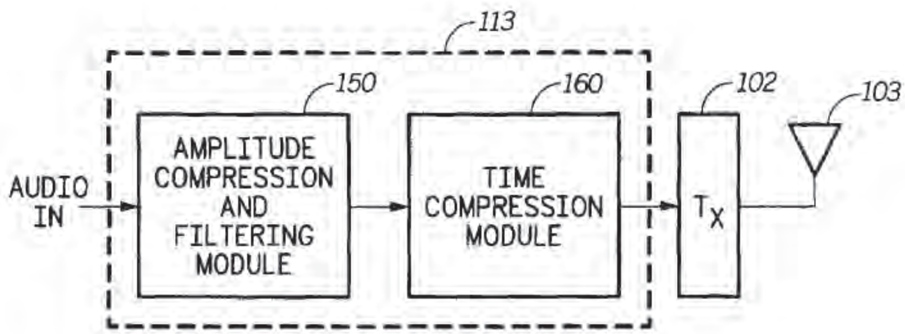

113
150
160
102
103
AUDIO IN
AMPLITUDE COMPRESSION AND FILTERING MODULE
TIME COMPRESSION MODULE
T_X

FIG. 2

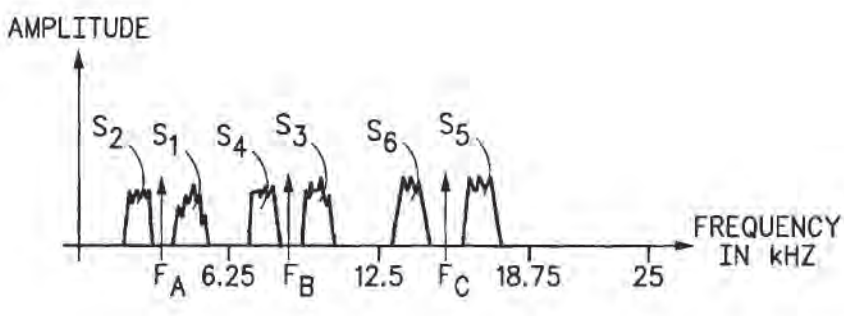
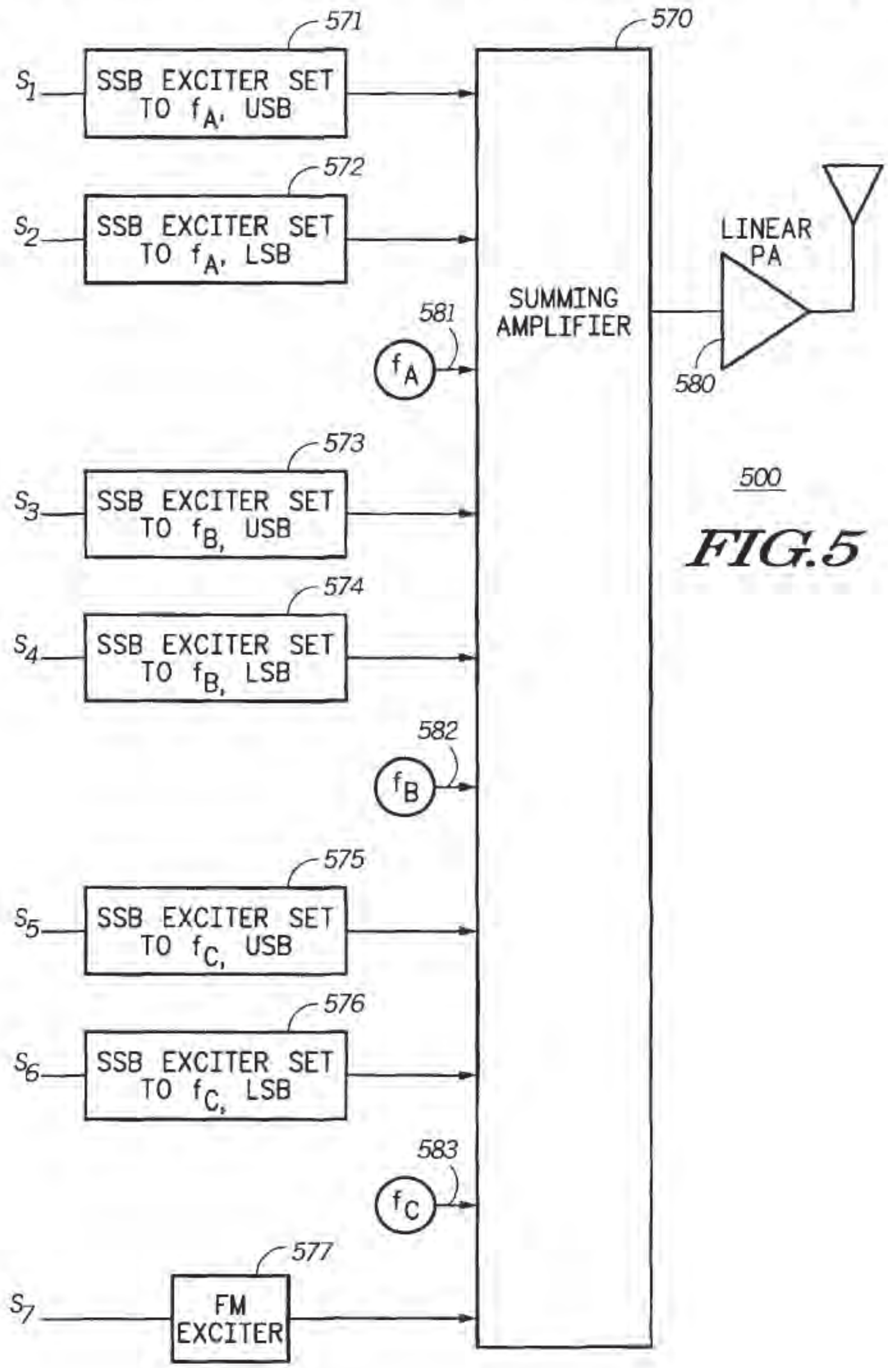
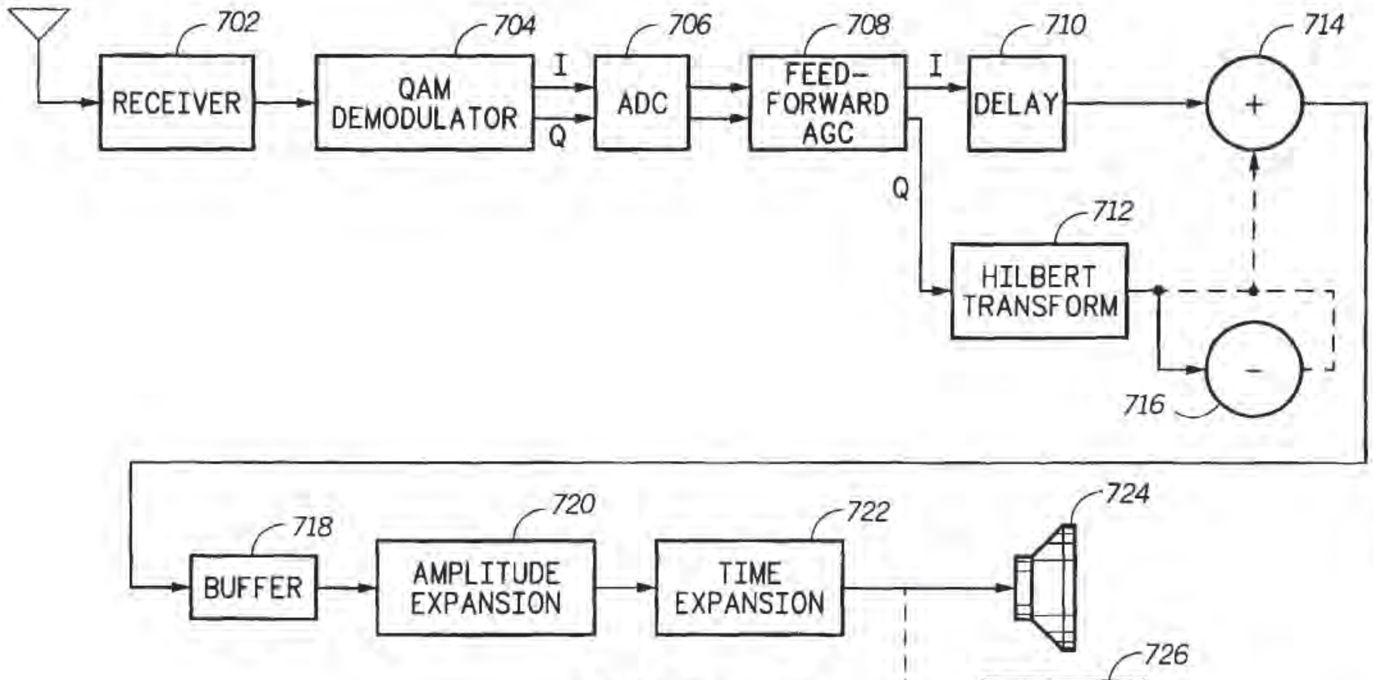

AMPLITUDE
S_2 S_1 S_4 S_3 S_6 S_5
FREQUENCY IN kHZ
F_A 6.25 F_B 12.5 F_C 18.75 25

FIG. 6


571
570
S_1
SSB EXCITER SET TO f_A, USB
572
S_2
SSB EXCITER SET TO f_A, LSB
LINEAR PA
SUMMING AMPLIFIER
581
f_A
580
573
S_3
SSB EXCITER SET TO f_B, USB
574
S_4
SSB EXCITER SET TO f_B, LSB
582
f_B
575
S_5
SSB EXCITER SET TO f_C, USB
576
S_6
SSB EXCITER SET TO f_C, LSB
583
f_C
577
S_7
FM EXCITER

500

FIG. 5



702
RECEIVER
704
QAM DEMODULATOR
I
Q
706
ADC
708
FEED-FORWARD AGC
I
Q
710
DELAY
714
+
712
HILBERT TRANSFORM
716
-
718
BUFFER
720
AMPLITUDE EXPANSION
722
TIME EXPANSION
724
726
COMPUTER

FIG. 7 700

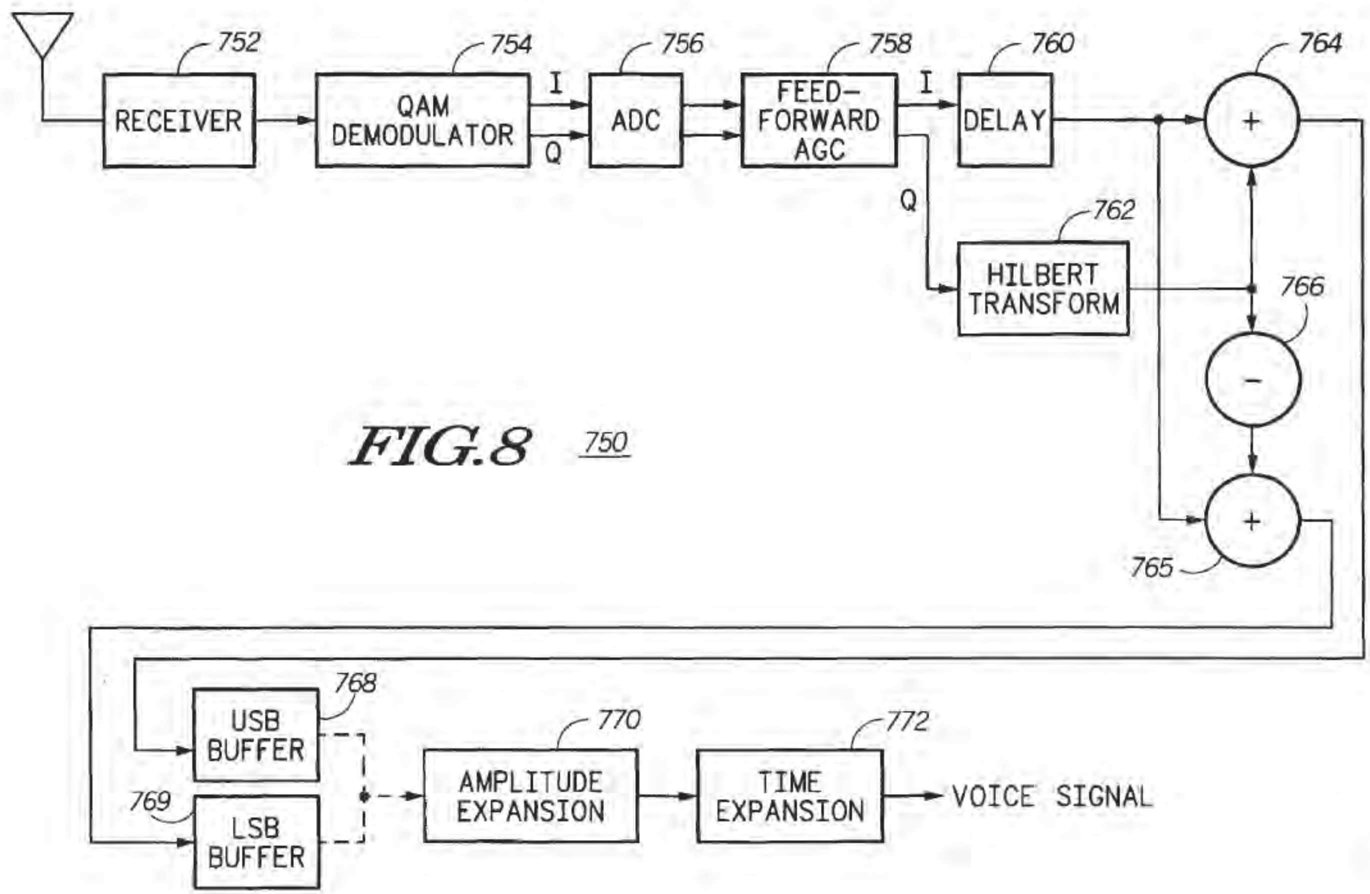
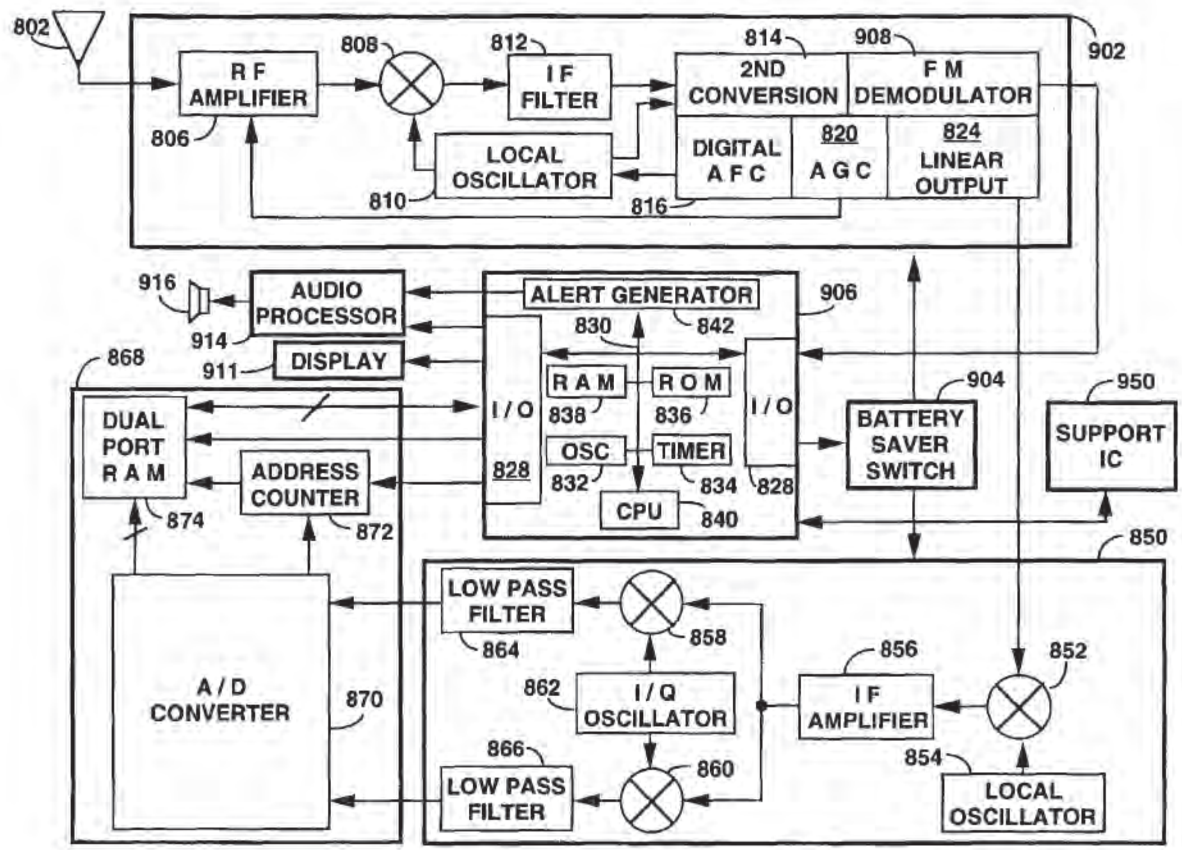

752
RECEIVER
754
QAM
DEMODULATOR
I
Q
756
ADC
758
FEED-
FORWARD
AGC
I
Q
760
DELAY
762
HILBERT
TRANSFORM
764
+
766
−
765
+
768
USB
BUFFER
769
LSB
BUFFER
770
AMPLITUDE
EXPANSION
772
TIME
EXPANSION
VOICE SIGNAL

FIG.8 750



802
808
812
814
908
902
R F AMPLIFIER
806
I F FILTER
2ND CONVERSION
F M DEMODULATOR
LOCAL OSCILLATOR
DIGITAL A F C
820
A G C
824
LINEAR OUTPUT
810
816
916
AUDIO PROCESSOR
ALERT GENERATOR
906
914
830
842
868
911
DISPLAY
R A M
R O M
I / O
I / O
904
950
DUAL PORT R A M
838
836
BATTERY SAVER SWITCH
SUPPORT IC
OSC
TIMER
ADDRESS COUNTER
828
832
834
828
CPU
874
872
840
850
LOW PASS FILTER
858
864
856
852
A / D CONVERTER
870
862
I / Q OSCILLATOR
I F AMPLIFIER
866
854
860
LOW PASS FILTER
LOCAL OSCILLATOR

FIG. 9 900

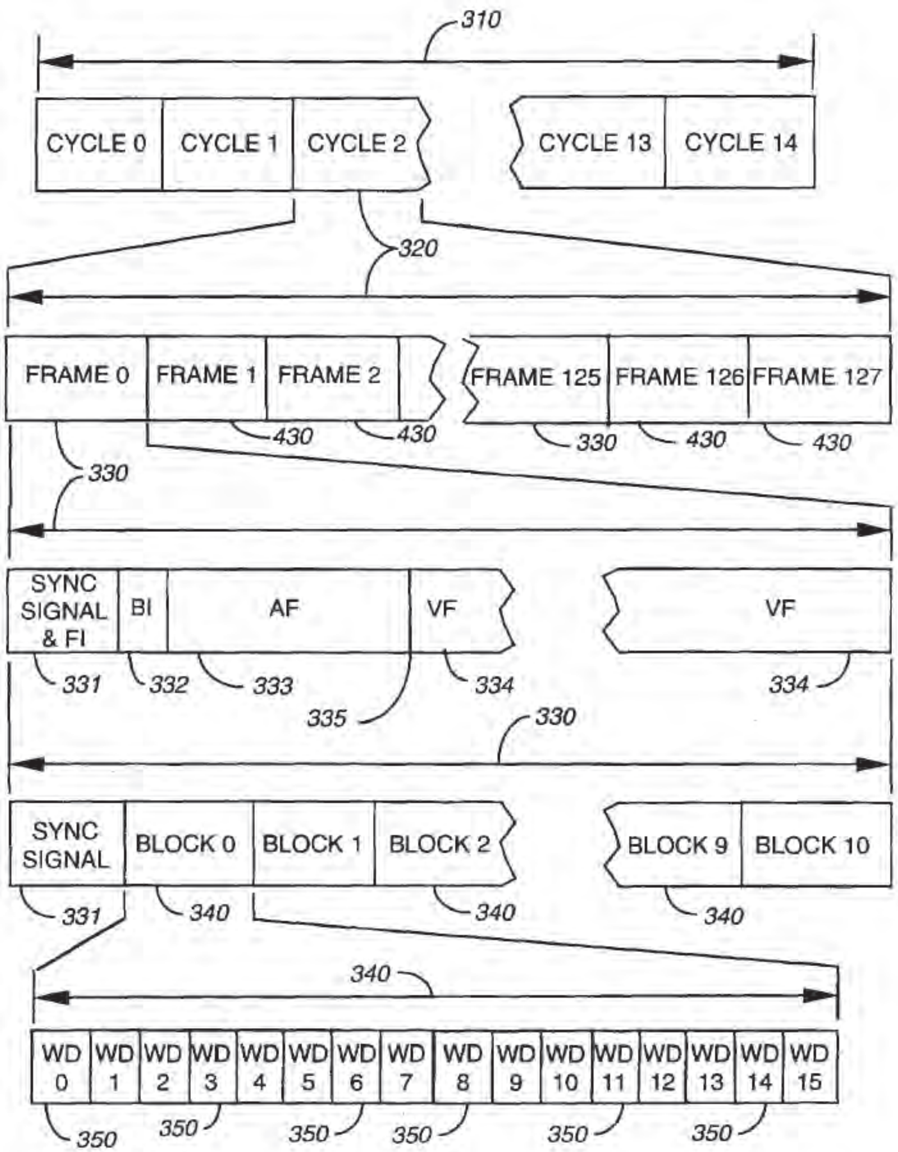

310
CYCLE 0
CYCLE 1
CYCLE 2
CYCLE 13
CYCLE 14
320
FRAME 0
FRAME 1
FRAME 2
FRAME 125
FRAME 126
FRAME 127
430
430
330
430
430
330
SYNC SIGNAL & FI
BI
AF
VF
VF
331
332
333
334
334
335
330
SYNC SIGNAL
BLOCK 0
BLOCK 1
BLOCK 2
BLOCK 9
BLOCK 10
331
340
340
340
340
WD 0
WD 1
WD 2
WD 3
WD 4
WD 5
WD 6
WD 7
WD 8
WD 9
WD 10
WD 11
WD 12
WD 13
WD 14
WD 15
350
350
350
350
350
350

FIG.10

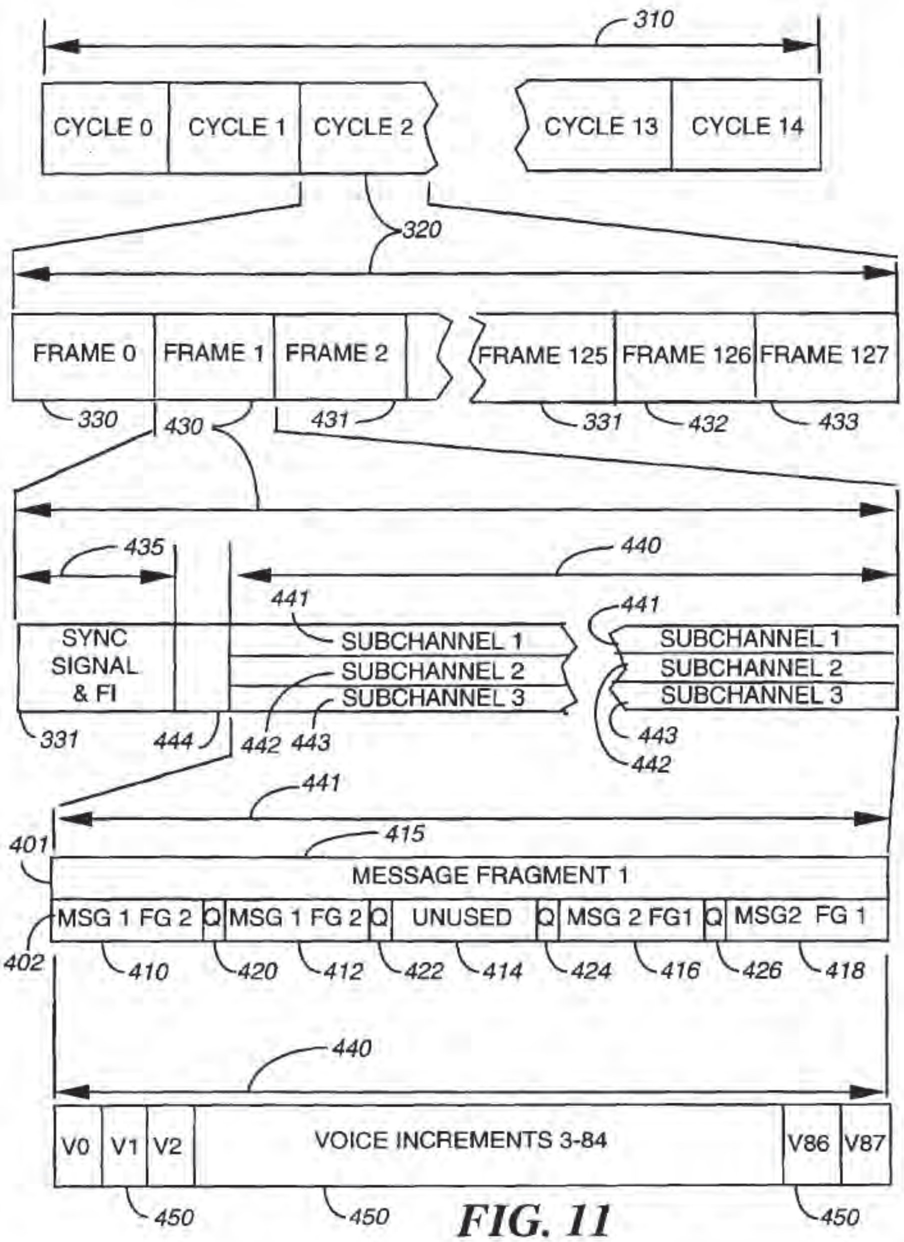

310
CYCLE 0
CYCLE 1
CYCLE 2
CYCLE 13
CYCLE 14
320
FRAME 0
FRAME 1
FRAME 2
FRAME 125
FRAME 126
FRAME 127
330
430
431
331
432
433
435
440
441
SYNC SIGNAL & FI
SUBCHANNEL 1
SUBCHANNEL 2
SUBCHANNEL 3
331
444
442
443
415
401
MESSAGE FRAGMENT 1
402
MSG 1 FG 2
Q
MSG 1 FG 2
Q
UNUSED
Q
MSG 2 FG1
Q
MSG2 FG 1
410
420
412
422
414
424
416
426
418
V0
V1
V2
VOICE INCREMENTS 3-84
V86
V87
450

FIG. 11

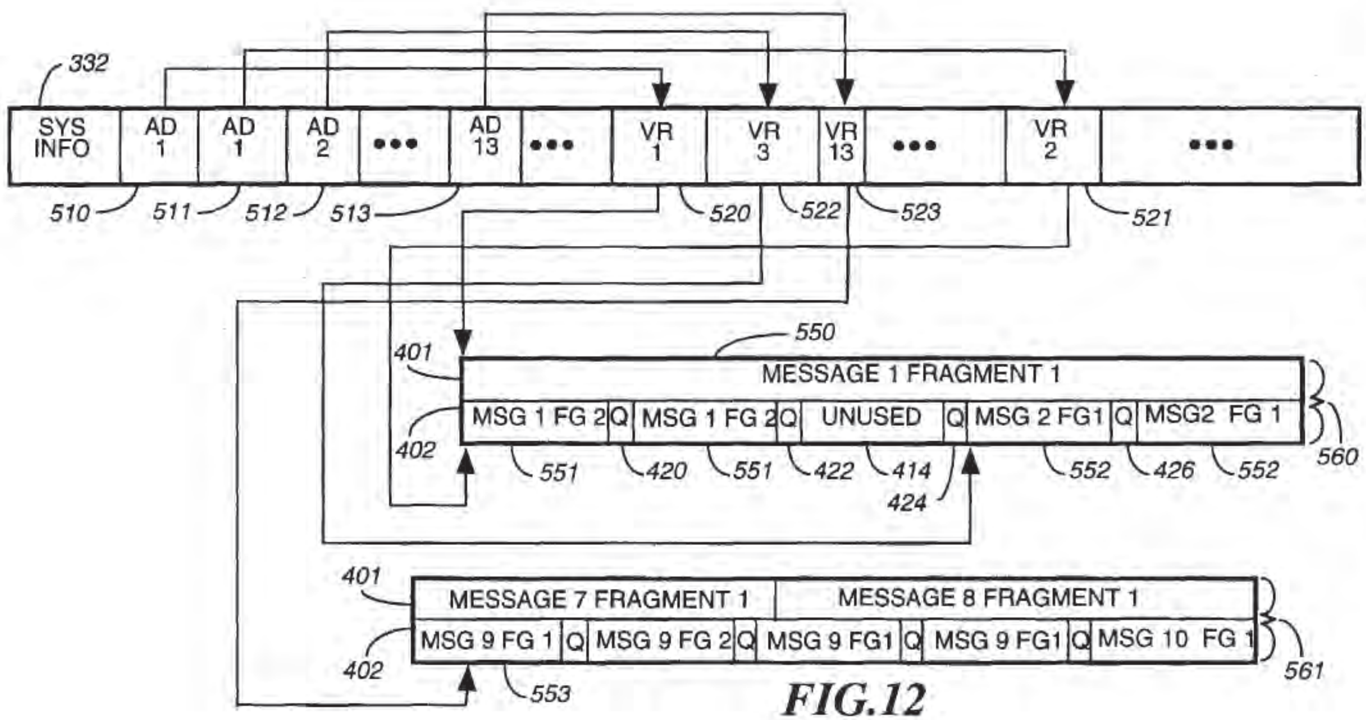

332
SYS INFO
AD 1
AD 1
AD 2
AD 13
VR 1
VR 3
VR 13
VR 2
510
511
512
513
520
522
523
521
550
401
MESSAGE 1 FRAGMENT 1
402
MSG 1 FG 2
Q
MSG 1 FG 2
Q
UNUSED
Q
MSG 2 FG1
Q
MSG2 FG 1
560
551
420
551
422
414
424
552
426
552
401
MESSAGE 7 FRAGMENT 1
MESSAGE 8 FRAGMENT 1
402
MSG 9 FG 1
Q
MSG 9 FG 2
Q
MSG 9 FG1
Q
MSG 9 FG1
Q
MSG 10 FG 1
561
553

FIG.12

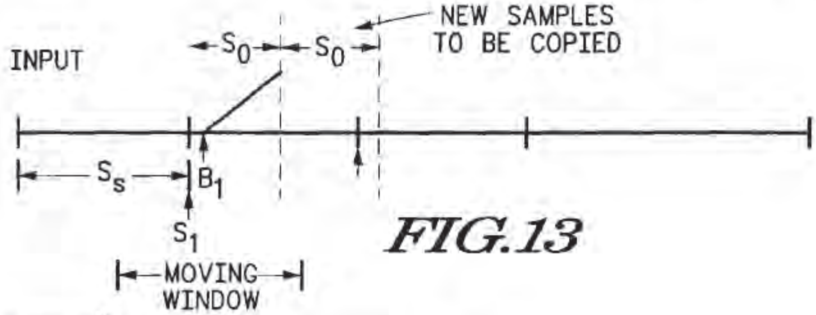


FIG. 13

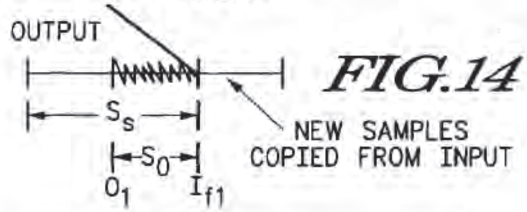


FIG. 14

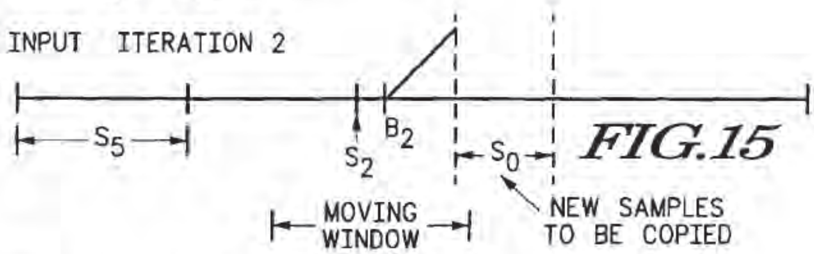


FIG. 15

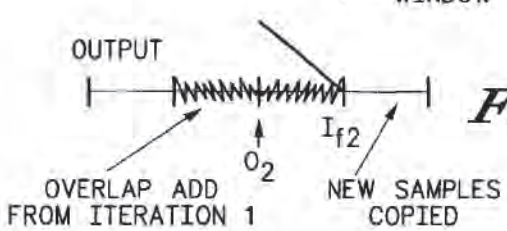


FIG. 16

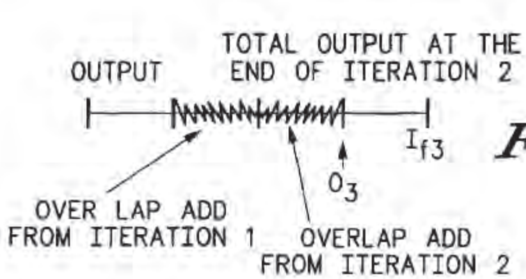


FIG. 17

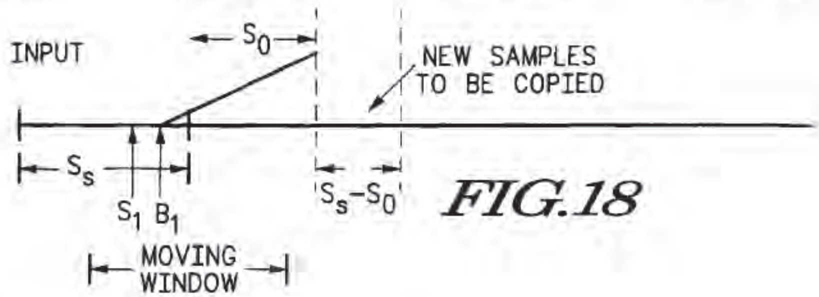


FIG. 18

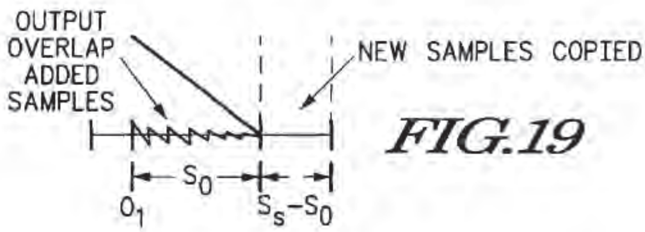


FIG. 19

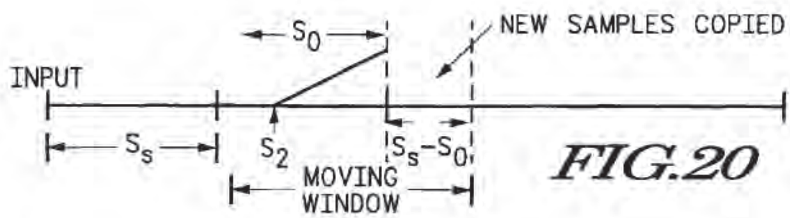


FIG. 20

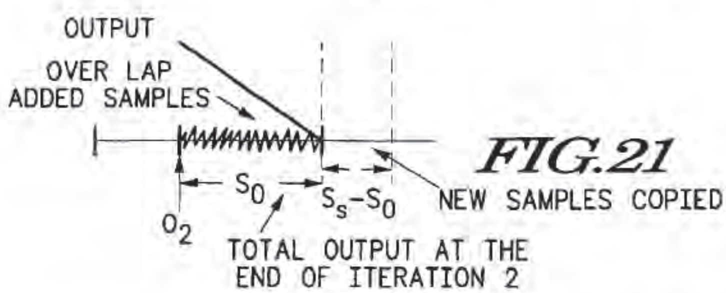


FIG. 21

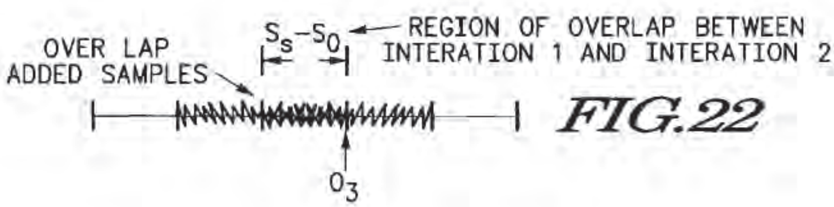


FIG. 22

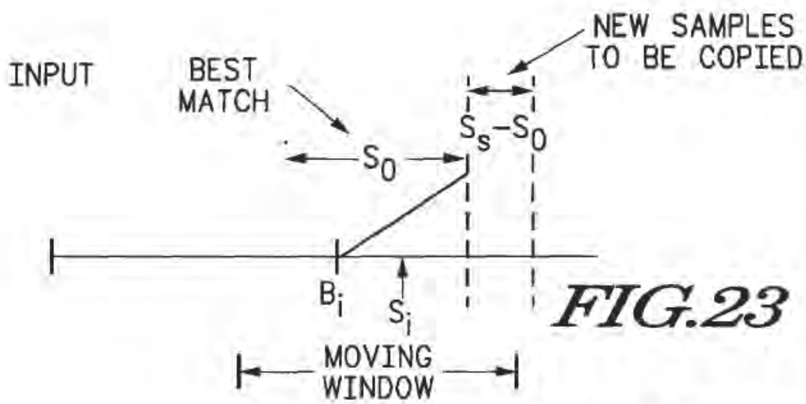


FIG.23

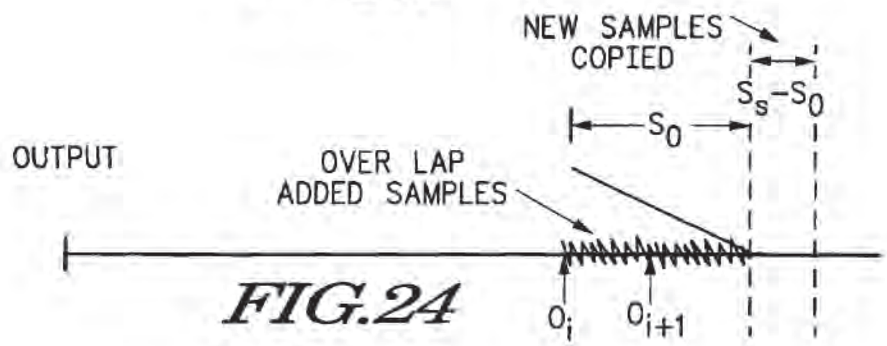


FIG.24

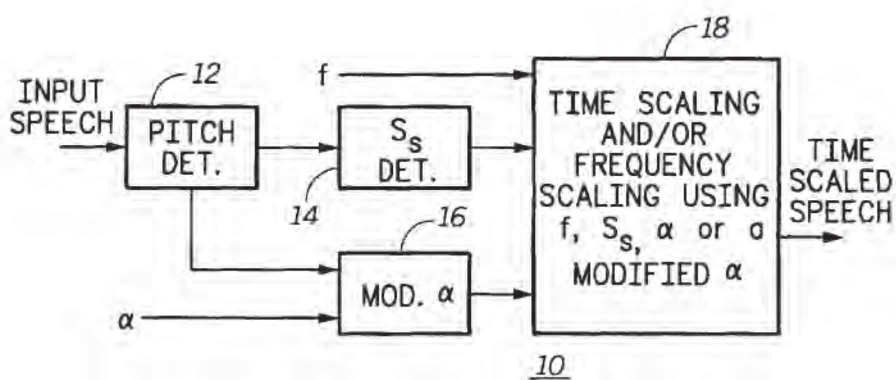


FIG.25

VOICE COMPRESSION METHOD AND APPARATUS IN A COMMUNICATION SYSTEM

This is a continuation of application Ser. No. 08/395,747 filed Feb. 28, 1995, now abandoned.

TECHNICAL FIELD

This invention relates generally to voice compression techniques, and more particularly a method and apparatus of voice compression using efficient bandwidth utilization and time compression techniques.

BACKGROUND

Voice message paging is not economically feasible for large paging systems with current technology. The air time required for a voice page is much more than that required for a tone, numeric or alphanumeric page. With current technology, voice paging service would be economically prohibitive in comparison to tone, numeric or alphanumeric paging with less than ideal voice quality reproduction. Another constraint in limiting voice message paging is the bandwidth and the present methods of utilizing the bandwidth of paging channels. In comparison, the growth of alphanumeric paging has been constrained by the limited access to a keyboard input device for sending alphanumeric messages to a paging terminal, either in the form of a personal keyboard or a call to an operator center. A voice system overcomes these entry issues since a caller can simply pick up a telephone, dial access numbers, and speak a message. Further, none of the present voice paging systems take advantage of Motorola's new high speed paging protocol structure, also known as FLEX™.

Existing voice paging systems lack many of the FLEX™ protocol advantages including high battery saving ratios, multiple channel scanning capability, mixing of modes such as voice with data, acknowledge-back paging (allowing for return receipts to the calling party), location finding capability, system and frequency reuse, particularly in large metropolitan areas, and range extension through selective re-transmission of missed message portions.

With respect to the aspect of paging involving time-scaling of voice signals and to other applications such as dictation and voice mail, current methods of time-scaling lack the ideal combinations of providing adequate speech quality and flexibility that allows a designer to optimize the application within the constraints given. Thus, there exists a need for a voice communication system that is economically feasible and flexible in allowing optimization within a given configuration, and more particularly with respect to paging applications, that further retains many of the advantages of Motorola's FLEX™ protocol.

SUMMARY OF THE INVENTION

In one aspect, the present invention comprises a method for compressing a plurality of voice signals within a voice communication resource having a given bandwidth within a voice communication system. The method comprises the steps of subchanneling the voice communication resource while placing at least one of each of the plurality of voice signals on a subchannel and compressing the time of each of the voice signals within each of the subchannels, wherein these steps provide a compressed voice signal.

In another aspect of the present invention, a communication system using voice compression has at least one transmitter base station and a plurality of selective call receivers. The transmitter base station comprises an input device for receiving an audio signal, a processing device for compressing the audio signal using a time-scale compression technique and a single side band modulation technique to provide a processed signal and a quadrature amplitude modulator for the subsequent transmission of the processed signal. Each of the plurality of selective call receivers comprises a selective call receiver module for receiving the transmitted processed signal, a processing device for demodulating the received processed signal using a single side band demodulation technique and a time-scale expansion technique to provide a reconstructed signal, and an amplifier for amplifying the reconstructed signal into an reconstructed audio signal.

In another aspect of the present invention, a selective call receiver for receiving compressed voice signals, comprises a selective call receiver module for receiving a transmitted processed signal, a processing device for demodulating the received processed signal using a single side band demodulation technique and a time-scale expansion technique to provide a reconstructed signal, and an amplifier for amplifying the reconstructed signal into an reconstructed audio signal.

In yet another aspect of the present invention, a paging base station for transmitting selective call signals on a communication resource having a predetermined bandwidth, comprises, an input device for receiving a plurality of audio signals, a device for subchannelizing the communication resource into a predetermined number of subchannels, an amplitude compression and filtering module for each subchannel for compressing the amplitude of the respective audio signal and filtering the respective audio signal, a time compression module for compression of the time of the respective audio signal for each subchannel, and a quadrature amplitude modulator for the subsequent transmission of the processed signal.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a block diagram of a voice communication system in accordance with the present invention.

FIG. 2 is a block diagram of a base station transmitter in accordance with the present invention.

FIG. 3 is an expanded electrical block diagram of the base station transmitter in accordance with the present invention.

FIG. 4 is an expanded electrical block diagram of another base station transmitter in accordance with the present invention.

FIG. 5 is block diagram of a speech processing, encoding, and modulation portion of a base station transmitter in accordance with the present invention.

FIG. 6 is a spectrum analyzer output of a 6 single-sideband signal transmitter in accordance with the present invention.

FIG. 7 is an expanded electrical block diagram of a selective call receiver in accordance with the present invention.

FIG. 8 is an expanded electrical block diagram of another selective call receiver in accordance with present invention.

FIG. 9 is an expanded electrical block diagram of another selective call receiver in accordance with present invention.

FIG. 10 is a timing diagram showing the transmission format of an outbound signaling protocol in accordance with the present invention.

FIG. 11 is another timing diagram showing the transmission format of an outbound signaling protocol including details of a voice frame in accordance with the present invention.

FIG. 12 is another timing diagram illustrating a control frame and two analog frames of the outbound signaling protocol in accordance with the present invention.

FIGS. 13–17 illustrate timing diagrams for several iterations of the WSOLA time-scaling (compression) method in accordance with the present invention.

FIGS. 18–22 illustrate timing diagrams for several iterations of the WSOLA-SD time-scaling (compression) method in accordance with the present invention.

FIGS. 23–24 illustrate timing diagrams for iterations of the WSOLA-SD time-scaling (expansion) method in accordance with the present invention.

FIG. 25 illustrates a block diagram of the overall WSOLA-SD time scaling method in accordance with the present invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

Referring to FIG. 1, a communication system illustrative of the voice compression and expansion techniques of the present invention are shown in a block diagram of the selective call system **100** which comprises an input device for receiving an audio signal such as telephone **114** (or other input device such as a computer) from which voice based selective calls are initiated for transmission to selective call receivers in the system **100**. Each selective call entered through the telephone **114** typically comprises (a) a receiver address of at least one of the selective call receivers in the system and (b) a voice message. The initiated selective calls are typically provided to a transmitter base station or a selective call terminal **113** for formatting and queuing. Voice compression circuitry **101** of the terminal **113** serves to compress the time length of the provided voice message (the detailed operation of such voice compression circuitry **101** is discussed in the following description of FIGS. **2**, **3** and **4**). Preferably, the voice compression circuitry **101** includes a processing device for compressing the audio signal using a time-scaling technique and a single sideband modulation technique to provide a processed signal. The selective call is then input to the selective call transmitter **102** where it is applied as modulation to a radio frequency signal which is sent over the air through an antenna **103**. Preferably, the transmitter is a quadrature amplitude modulation transmitter for transmitting the processed signal.

An antenna **104** within a selective call receiver **112** receives the modulated, transmitted radio frequency signal and inputs it to a selective call receiver module or radio frequency receiver module **105** for receiving the processed signal or radio frequency signal, where the radio frequency signal is demodulated and the receiver address and the compressed voice message modulation are recovered. The compressed voice message is then provided to an analog to digital converter (A/D) **115**. Preferably, the selective call receiver **112** includes a processing device for demodulating the received processed signal using a single sideband demodulation technique and a time-scaling expansion technique to provide a reconstructed signal. The compressed voice message is then provided to a voice expansion circuit **106** where the time length of the voice message is preferably expanded to the desired value (the detailed operation of such voice expansion circuitry **106** used in the present invention is discussed in the following description of FIGS. **7** and **8**). The voice message is then provided to an amplifier such as audio amplifier **108** for the purpose of amplifying it to a reconstructed audio signal.

The demodulated receiver address is supplied from the radio frequency receiver **105** to a decoder **107**. If the

receiver address matches any of the receiver addresses stored in the decoder **107**, an alert **111** is optionally activated, providing a brief sensory indication to the user of the selective call receiver **112** that a selective call has been received. The brief sensory indication may comprise an audible signal, a tactile signal such as a vibration, or a visual signal such as a light, or a combination thereof. The amplified voice message is then furnished from the audio amplifier **108** to an audio loudspeaker within the alert **111** for message announcement and review by the user.

The decoder **107** may comprise a memory in which the received voice messages can be stored and recalled repeatedly for review by actuation of one or more controls **110**.

In another aspect of the invention, portions of FIG. **1** can be equally interpreted as part of a dictation device, voice mail system, answering machine, or sound track editing device for example. By removing the wireless aspects of the system **100** including the removal of selective call transmitter **102** and radio frequency receiver **105**, the system can be optionally hardwired from the voice compression circuitry **101** to the voice expansion circuitry **106** through the A/D **115** as shown with the dashed line. Thus, in a voice mail, answering machine, sound track editing or dictation system, an input device **114** would supply an acoustic input signal such as a speech signal to the terminal **113** having the voice compression circuitry **101**. The voice expansion circuitry **106** and controls **110** would supply the means of listening and manipulating to the output speech signal in a voice mail, answering machine, dictation, sound track editing or other applicable system. This invention clearly contemplates that the time-scaling techniques of the claimed invention has many other applications besides paging. The paging example disclosed herein is merely illustrative of one of those applications.

Now referring to FIG. **2**, there is shown a block diagram of a paging transmitter **102** and terminal **113** including an amplitude compression and filtering module **150** coupled to a time compression module **160** which is coupled to the selective call transmitter **102** and which transmits messages using aerial or antenna **103**. Referring to FIGS. **3** and **4**, a lower level block diagram of the block diagram of FIG. **2** is shown.

Please keep in mind that this compressed voice paging system is highly bandwidth efficient and intended to support typically 6 to 30 voice messages per 25 kHz channel using the basic concepts of quadrature amplitude (QAM) or single-side band (SSB) modulation and time scaling of speech signals. Preferably, in a first embodiment and also referring to FIG. **6**, the compressed voice channel or voice communication resource consists of 3 sub-channels that are separated by 6250 Hz. Each sub-channel consists of two side-bands and a pilot carrier. Each of these two side-bands may have the same message in a first method or separate speech messages on each sideband or a single message split between the upper and lower sidebands in a second method. The single sub-channel has a bandwidth of substantially 6250 Hz with each side-band occupying a bandwidth of substantially 3125 Hz. The actual speech bandwidth is substantially 300–2800 Hz. Alternatively, the quadrature amplitude modulation may be used where the two independent signals are transmitted directly via I and Q components of the signal to form each sub-channel signal. The bandwidth required for transmission is the same in the QAM and SSB cases.

Note that modules **150** and **160** in FIG. **2** can be repeated for use by each different voice signal (up to 6 times in 25

KHz wide channels and up to 14 times in 50 KHz wide channels) to allow for the efficient and simultaneous transmission of (up to 6 in examples shown) voice messages. They can all then be summed at a summing device (not shown, but see FIG. 5) and preferably processed as a composite signal in **102**. A separate signal (not shown) contains the FM modulation of the FLEX™ protocol (as will be described later) which may optionally be generated in software or as a hardware FM signal exciter.

Preferably, in the examples shown herein, an incoming speech message is received by the terminal **113**. The present system preferably uses a time-scaling scheme or technique to achieve the required compression. The preferred compression technique used in the present invention requires certain parameters specific to the incoming message to provide an optimum quality. Preferably, the technique of time-scale compression processes the speech signal into a signal having the same bandwidth characteristics as uncompressed speech. (Once these parameters are computed, speech is compressed using the desired time-scaling compression technique). This time-scaled compressed speech is then encoded using a digital coder to reduce the number of bits required to be distributed to the transmitters. In the case of a paging system, the encoded speech distributed to the transmitters of multiple simulcasting sites in a simulcasting paging system would need to be decoded once again for further processing such as amplitude compression. Amplitude compression of the incoming speech signals (preferably using a syllabic compander) is used at the transmitter to give protection against channel impairments.

A time scaling technique known as Waveform Similarity based Overlap-Add technique or WSOLA encodes speech into an analog signal having the same bandwidth characteristics as uncompressed speech. This property of WSOLA allows it to be combined with SSB or QAM modulation such that the overall compression achieved is the product of the bandwidth compression ratio of multiple QAM or SSB subchannels (in our example, 6 voice channels) and the time compression ratio of WSOLA (typically between 1 and 5). In the present invention, a modified version of WSOLA, later described and referred to as "WSOLA-SD" is used. WSOLA-SD retains the compatibility characteristics of WSOLA that allows the combination with SSB or QAM modulation.

Preferably, an Adaptive Differential Pulse Coded Modulation coder (ADPCM) is used to encode the speech into data that is subsequently distributed to the transmitters. At the transmitter, the digital data is decoded to obtain WSOLA-SD compressed speech which is then amplitude companded to provide protection against channel noise. This signal is Hilbert transformed to obtain a single-sideband signal. Alternatively, the signal is quadrature modulated to obtain a QAM signal. A pilot carrier is then added to the signal and the final signal is interpolated, preferably, to a 16 kHz sampling rate and converted to analog. This is then modulated and transmitted.

The present invention can operate as a mixed-mode (voice or digital) one or two way communications system for delivering analog voice and/or digital messages to selective call receiver units on a forward channel (outbound from the base transmitter) and for receiving acknowledgments from the same selective call receiver units which additionally have optional transmitters (on an optional reverse channel (inbound to a base receiver). The system of the present invention preferably utilizes a synchronous frame structure similar to FLEX™ (a high speed paging protocol by Motorola, Inc. and subject of U.S. Pat. No. 5,282,205, which is hereby incorporated by reference) on the forward channel for both addressing and voice messaging. Two types of frames are used: control frames and voice frames. The control frames are preferably used for addressing and delivery of digital data to selective call receivers in the form of portable voice units (PVU's). The voice frames are used for delivering analog voice messages to the PVU's. Both types of frames are identical in length to standard FLEX™ frames and both frames begin with the standard FLEX™ synchronization. These two types of frames are time multiplexed on a single forward channel. The frame structure for the present invention will be discuss in greater detail later on with regard to FIGS. **10**, **11** and **12**.

With regard to modulation, two types of modulation are preferably used on the forward channel of the present invention: Digital FM (2-level and 4-level FSK) and AM (SSB or QAM with pilot carrier). Digital FM modulation is used for the sync portions of both types of frames, and for the address and data fields of the control frames. AM modulation (each sideband maybe used independently or combined together in a single message) is used in the voice message field of the voice frames. The digital FM portions of the transmission support 6400 BPS (3200 Baud symbols) signaling. The AM portions of the transmissions support band limited voice (2800 Hz) and require 6.25 KHz for a pair of voice signals. The protocol, as will be shown later, takes advantage of the reduced AM bandwidth by subdividing a full channel into 6.25 KHz subchannels, and by using each subchannel and the AM sidebands for independent messages.

Voice System of the present invention is preferably designed to operate on either 25 KHz or 50 KHz forward channels, but other size spectrum is certainly within contemplation of the present invention. A 25 KHz forward channel supports a single FM control signal during control frames, and up to 3 AM subchannels (6 independent signals) during the message portion of voice frames. A 50 KHz forward channel supports two FM control signals operated in time lock during control frames, and up to 7 AM subchannels (14 independent signals) during the message portion of voice frames. Of course, other configurations using different size bandwidths and numbers of subchannels and signals are contemplated within the present invention. The examples disclosed herein are merely illustrative and indicative of the potential broad scope of the claims herein.

In addition to the spectrum efficiency achieved through modulation and sub-channelization of the spectrum, the present invention, in another embodiment, can utilize a speaker dependent voice compression technique that time scales the speech by a factor of 1 to 5 times. By using both AM sidebands (alternatively, the 2 QAM components) of a subchannel for different portions of the same message or different messages, the overall compression factor per subchannel is 2 to 10 times. Voice quality will typically decrease with an increasing time-compression factor. The compression technique preferably used in the voice system of the present invention is a modified form of a known time-scaling technique known as Waveform Similarity based Overlap-Add technique (WSOLA) as previously mentioned. The modified form of WSOLA is dependent upon the particular speaker or speech used, hence the name "WSOLA-SD" for "WSOLA-Speaker dependent", which will be discussed later on.

Operation of the present invention is enhanced when a reverse (inbound to the base receiver) channel is available. The frequency division simplex mode of operation is one inbound operating mode supported. (U.S. Pat. Nos. 4,875, 038 and 4,882,579, both assigned to assignee of the present invention, Motorola, Inc., illustrate the use of multiple acknowledge signals on an inbound channel and are incorporated herein by reference). In frequency division simplex, a separate dedicated channel (usually paired with the outbound channel) is provided for inbound transmissions. Inbound data rates of 800 to 9600 BPS are contemplated within a channel bandwidth of 12.5 KHz.

The system of the present invention can be operated in one of several modes depending on the availability of a reverse channel. When no reverse channel is available, the system is preferably operated in simulcast mode for both addressing and voice messaging. When a reverse channel is provided, the system can be operated in a targeted message mode whereby the messages are broadcast only on a single or a subset of transmitters located near the portable voice unit. The targeted message mode is characterized by simulcast addressing to locate the portable voice unit, the portable voice unit's response on the reverse channel provides the location, followed by a localized message transmission to the portable voice unit. The targeted message mode of operation is advantageous in that it provides the opportunity for subchannel reuse; and consequently, this mode of operation can lead to increased system capacity in many large systems.

FIG. 3 illustrates a block diagram of a first embodiment of a transmitter **300** in accordance with the present invention. An analog speech signal is input to an anti-aliasing low pass filter **301** which strongly attenuates all frequencies above one-half the sampling rate of an analog-to-digital converter (ADC) **303** which is further coupled to the filter **301**. The ADC **303** preferably converts the analog speech signal to a digital signal so that further signal processing can be done using digital processing techniques. Digital processing is the preferred method, but the same functions could also be performed with analog techniques or a combination of analog and digital techniques.

A band pass filter **305** coupled to the ADC **303** strongly attenuates frequencies below and above its cutoff frequencies. The lower cutoff frequency is preferably 300 Hz which allows the significant speech frequencies to pass, but attenuates lower frequencies which would interfere with a pilot carrier. The upper cutoff frequency is preferably 2800 Hz which allows the significant speech frequencies to pass but attenuates higher frequencies which would interfere with adjacent transmission channels. An automatic gain control (AGC) block **307** preferably coupled to the filter **305** equalizes the volume level of different voices.

A time compression block **309** preferably coupled to the AGC block **307** shortens the time required for transmission of the speech signal while maintaining essentially the same signal spectrum as at the output of the bandpass filter **305**. The time compression method is preferably WSOLA-SD (as will be explained later on), but other methods could be used. An amplitude compression block **311**, and the corresponding amplitude expansion block **720** in a receiver **700** (FIG. 7), form a companding device which is well known to increase the apparent signal-to-noise ratio of the received speech. The companding ratio is preferably 2 to 1 in decibels, but other ratios could be used in accordance with the present invention. In the particular instance of a communication system such as a paging system, the devices 301–309 may be included in a paging terminal (**113** of FIG. 1) and the remaining components in FIG. 3 could constitute a paging transmitter (**102** of FIG. 1). In such a case, there would typically be a digital link between the paging terminal and paging transmitter. For instance, the signal after block **309** could be encoded using a pulse code modulation (PCM) technique and then subsequently decoded using PCM to reduce the number of bits transferred between the paging terminal and paging transmitter.

In any event, a second band pass filter **308** coupled to the amplitude compression block **311** strongly attenuates frequencies below and above its cutoff frequencies to remove any spurious frequency components generated by the AGC **307**, the time compression block **309** or the amplitude compression block **311**. The lower cutoff frequency is preferably 300 Hz which allows the significant speech frequencies to pass, but attenuates lower frequencies which would interfere with the pilot carrier. The upper cutoff frequency is preferably 2800 Hz which allows the significant speech frequencies to pass but attenuates higher frequencies which would interfere with adjacent transmission channels.

The time compressed speech samples are preferably stored in a buffer **313** until an entire speech message has been processed. This allows the time compressed speech message to then be transmitted as a whole. This buffering method is preferably used for paging service (which is typically a non real time service). Other buffering methods may be preferable for other applications. For example, for an application involving two-way real time conversation, the delay caused by this type of buffering could be intolerable. In that case it would be preferable to interleave small segments of several conversations. For example, if the time compression ratio is 3:1, then 3 real time speech signals could be transmitted via a single channel. The 3 transmissions could be interleaved on the channel in 150 millisecond bursts and the resulting delays would not be objectionable. The time compressed speech signal from the buffer **313** is applied to both to a Hilbert transform filter **323** and to a time delay block **315** which has the same delay as the Hilbert transform filter, but does not otherwise affect the signal.

The output of the time delay block **315** (through the summing circuit **317**) and the Hilbert transform filter **323** form, respectively, the in-phase (I) and quadrature (Q) components of an upper sideband (USB) single sideband (SSB) signal. The output of the time delay and the negative (**325**) of the Hilbert transform filter form, respectively, the in-phase (I) and quadrature (Q) components of a lower sideband (LSB) single sideband signal. Thus the transmission may be on either the upper or lower sideband, as indicated by the dotted connection.

While the upper sideband is used to transmit one time compressed speech signal, the lower sideband can be used to simultaneously transmit a second time compressed speech signal by using another similar transmitter operating on the lower sideband. SSB is the preferred modulation method because of efficient use of transmission bandwidth and resistance to crosstalk. Double sideband Amplitude Modulation (AM) or frequency modulation (FM) could be used, but would require at least twice the bandwidth for transmission. It is also possible to transmit one time compressed speech signal directly via the I component and a second time compressed speech signal directly via the Q component, however, in the present embodiment this method is subject to crosstalk between the two signals when multipath reception occurs at the receiver.

A direct current (DC) signal is added to the I component of the signal to generate the pilot carrier, which is transmitted along with the signal and used by the receiver (**700**) to substantially cancel the effects of gain and phase variations or fading in the transmission channel. The I and Q components of the signal are convened to analog form by digital-to-analog converters (DAC) **319** and **327** respectively. The two signals are then filtered by low pass reconstruction filters **321** and **329** respectively to remove spurious frequency components resulting from the digital-to-analog conversion process. A quadrature amplitude modulation (QAM) modulator **333** modulates the I and Q signals onto a radio frequency (RF) carrier at low power level. Other modulation methods, e.g. direct digital synthesis of the modulated signal would accomplish the same purpose as the DACs (**319** and **327**), reconstruction filters (**321** and **329**), and QAM modulator **333**. Finally, a linear RF power amplifier **335** amplifies the modulated RF signal to the desired power level, typically 50 watts or more. Then, the output of the RF power amplifier **335** is coupled to the transmitting antenna. Other variations can produce essentially the same results. For example, the amplitude compression could be performed before the time compression, or omitted altogether and the device would still perform essentially the same function.

FIG. **4** illustrates a block diagram of a second embodiment of a transmitter **400** in accordance with the present invention. In FIG. **4**, both the upper and lower sidebands are used to simultaneously transmit different portions of the same time compressed signal. The transmitter **400** preferably includes an anti-alias filter **404**, an ADC **403**, a bandpass filter **405**, an AGC **407**, a time compression block **409**, an amplitude compression block **411**, and a bandpass filter **408** coupled and configured as in FIG. **3**. Operation of the transmitter of FIG. **4** is the same as in FIG. **3** until an entire speech message has been processed and stored in a buffer **413**. The time compressed speech samples stored in the buffer **413** are then divided to be transmitted on either the upper or lower sideband. Preferably, the first half of the time compressed speech message is transmitted via one sideband and the second half of the time compressed speech message is transmitted via the other sideband (or alternatively on each of the I and Q components directly).

The first portion of time compressed speech signal from the buffer **413** is applied to both a first Hilbert transform filter **423** and to a first time delay block **415** which has the same delay as the Hilbert transform filter **423** but does not otherwise affect the signal. The output of the first time delay (through summing circuit **417**) and the first Hilbert transform filter **423** (through summing circuit **465**) are In-Phase (I) and Quadrature Phase (Q) signal components which, when coupled to I and Q inputs of the QAM modulator, generate upper sideband signal having information only from the first portion of time compressed speech samples. The second time compressed speech signal from the buffer **413** is applied to both a second Hilbert transform filter **461** and to a second time delay block **457** which has the same delay as the Hilbert transform filter **461** but does not otherwise affect the signal. The output of the second time delay (through summing circuits **459** and **417**) and the negative (**463**) of the output of the second Hilbert transform filter **461** (and again, through summing circuit **465**) are In-Phase (I) and Quadrature Phase (Q) signal components which, when coupled to I and Q inputs of the QAM modulator, generate upper sideband signal having information only from the second portion of time compressed speech samples. The I components of the upper and lower sideband signals are added with a DC pilot carrier component (through summing circuit **459**) to form a composite I component for transmission. The Q components of the upper and lower sideband signals are added (through summing circuit **465**) to form a composite Q component for transmission. It will be appreciated that elements **415**, **423**, **457**, **461**, **417**, **459**, **463**, **465**, **419**, **427**, **421**, and **429** form a preprocessor which generates preprocessed I and Q signal components, which when coupled to the QAM modulator **453** generate the low level subchannel signal with a subcarrier F_A, having two single sideband signals, which have independent information on each sideband.

The transmitter **400** further comprises DACs **419** and **427**, reconstruction filters **421** and **429**, QAM modulator **433**, and RF power amplifier **455** arranged and constructed as described in FIG. **3**. Operation of the rest of the transmitter of FIG. **4** is the same as in FIG. **3**.

Preferably, in both transmitters **300** and **400** of FIGS. **3** and **4** respectively, only the anti-alias filters, the reconstruction filters, the RF power amplifier and optionally the Analog to Digital converter and digital to analog converters are separate hardware components. The remainder of the devices can preferably be incorporated into software which could be run on a processor, preferably a digital signal processor.

FIG. **7** illustrates a block diagram of a receiver **700** which preferably operates in conjunction with the transmitter **300** of FIG. **3** in accordance with the present invention. A receiving antenna is coupled to a receiver module **702**. The receiver module **702** includes conventional receiver elements, such as RF amplifier, mixer, bandpass filter, and intermediate frequency (IF) amplifier (not shown). A QAM demodulator **704** detects the I and Q components of the received signal. An analog-to-digital converter (ADC) **706** converts the I and Q components to digital form for further processing. Digital processing is the preferred method, but the same functions could also be performed with analog techniques or a combination of analog and digital techniques. Other methods of demodulation, e.g., a sigma-delta converter, or direct digital demodulation, would accomplish the same purpose as the QAM demodulator **704** and ADC **706**.

A feedforward automatic gain control (AGC) block **708** uses the pilot carrier, transmitted along with the time compressed speech signal, as a phase and amplitude reference signal to substantially cancel the effects of amplitude and phase distortions occurring in the transmission channel. The outputs of the feedforward automatic gain control are corrected I and Q components of the received signal. The corrected Q component is applied to a Hilbert transform filter **712**, and the corrected I component is applied to a time delay block **710** which has the same delay as the Hilbert transform filter **712** but does not otherwise affect the signal.

If the time compressed speech signal was transmitted on the upper sideband, the output of the Hilbert transform filter **712** is added (through summing circuit **714**) to the output of the time delay block **710** to produce the recovered time compressed speech signal. If the time compressed speech signal was transmitted on the lower sideband, the output of the Hilbert transform filter **712** is subtracted (**716**) from the output of the time delay block **710** to produce the recovered time compressed speech signal. The recovered time compressed speech signal is preferably stored in a buffer **718** until an entire message has been received. Other buffering methods are also possible. (See the discussion with FIG. **3**.)

An amplitude expansion block **720** works in conjunction with the amplitude compression block **311** of FIG. **3** to perform the companding function. A time expansion block **722** works in conjunction with the time compression block **309** of FIG. **3** and preferably reconstructs the speech into its natural time frame for an audio output through transducer **724** or other time frames as other applications may suggest.

One application could optionally include the transfer of digitized voice to a computing device 726, where the receiver-to-computer interface can be a PCMCIA or RS-232 interface or any number of interfaces known in the art. The time compression method is preferably WSOLA-SD, but other methods could be used, so long as complementary methods are used in the transmitter and receiver. Other variations in configuration can produce essentially the same results. For example, the amplitude compression could be performed after the time compression, or omitted altogether and the device would still perform essentially the same function.

FIG. 8 illustrates a block diagram of a receiver 750 which operates in conjunction with the transmitter of FIG. 400 in accordance with the present invention. The receiver of FIG. 8 comprises an antenna, receiver module 752, a QAM modulator 754, an ADC 756, a Feed-forward AGC 758, a time delay block 760, and a Hilbert transform filter 762 arranged and constructed as described in FIG. 7. Operation of the receiver of FIG. 8 is the same as FIG. 7, up to the output of the time delay block 760 and Hilbert transform filter 762. The output of the Hilbert transform filter 762 is added to the output of the time delay block 760 (through summing circuit 764) to produce the recovered time compressed speech signal corresponding to the first half of the speech message which was transmitted on the upper sideband. The output of the Hilbert transform filter 762 is subtracted (766) from the output of the time delay block 760 to produce the recovered time compressed speech signal corresponding to the second half of the speech message which was transmitted on the lower sideband.

The two recovered time compressed speech signals are stored in either respective upper sideband and lower sideband buffers 768 or 769 until the entire message has been received. Then, the signal corresponding to the first half of the message and the signal corresponding to the second half of the message are applied sequentially to the amplitude expansion block 770. An amplitude expansion block 770 works in conjunction with the amplitude compression block 411 of FIG. 4 to perform the companding function.

The operation of the rest of the receiver of FIG. 8 is the same as FIG. 7. A time expansion block 772 works in conjunction with the time compression block 409 of FIG. 4 and preferably reconstructs the speech into its natural time frame or other time frames as other applications may suggest or require. The time compression method is preferably WSOLA-SD, but other methods could be used, so long as complementary methods are used in the transmitter and receiver. Other configurations can produce essentially the same results. For example, the amplitude compression could be performed after the time compression, or omitted altogether and the device would still perform essentially the same function.

As with the implementation of the transmitters of FIGS. 3 and 4, many of the components in FIGS. 7 and 8 can be implemented in software including, but not limited to the AGCs, the single-sideband or QAM demodulators, summation circuits, the amplitude expansion blocks, and the time expansion blocks. All the other components are preferably implemented in hardware.

If the speech processing, encoding and modulation portion of the present invention were to be implemented into hardware, the implementation of FIG. 5 could be used. For instance, transmitter 500 of FIG. 5 would include a series of pairs of single-sideband exciters (571–576) set to the frequencies of their respective pilot carriers (581–583). Exciters 571–576 and pilot carriers 581–583 correspond to the separate voice processing paths. All these signals, including a signal from an FM signal exciter 577 (for the digital FM modulation used for the synchronization, address and data fields previously described) would be fed into a summing amplifier 570 which in turn is amplified by a linear amplifier 580 and subsequently transmitted. The low level output of FM exciter 577 is also linearly combined in summing amplifier 570. The composite output signal of summing amplifier 570 is amplified to the desired power level, usually 50 watts or more, by linear RF power amplifier 580. The output of linear RF power amplifier 580 is then coupled to the transmitting antenna.

Other means could be used to combine several subchannel signals. For example, the several digital baseband I and Q signals, obtained at the outputs of 417 and 465 in FIG. 4, could be translated in frequency to their respective subcarrier offset frequencies, combined in digital form, then converted to analog form for modulation onto the carrier frequency.

Referring to FIG. 9, there is shown another receiver unit 900 in accordance with the present invention. Receiver 900 additionally incorporates a means for detecting and decoding the FM modulated control signals that are used in the FLEX™ signaling protocol. Block 902 is the receiver front end and an FM back end. A digital automatic frequency controller (DAFC) and automatic gain controller (AGC) are incorporated into block 902. Block 906 includes the radio processor with a support chip 950 and Blocks 911, 914, and 916 include all the output devices. Block 904 is the battery saver or battery economy circuit which operates under control of the processor 906. Block 850 is the linear decoder followed by an analog-to-digital converter and random access memory (RAM) Block 868. The receiver Block 902 is preferably a modified FM receiver including the addition of a DAFC as described in U.S. Pat. No. 5,239,306 (which is assigned to the assignee of the present invention and which is hereby incorporated by reference herein), an AGC, and which provides for an intermediate frequency (IF) output at a point following most of the receiver gain but prior to the FM demodulator.

The same processor that controls Motorola's FLEX™ protocol compatible pagers would adequately handle all the protocol functions in the present invention including the address recognition and message decoding of an FM demodulated signal. Additionally, in response to an FM modulated address (and perhaps message pointer code words), the processor 906 initiates the operation of the analog-to-digital conversion and of the RAM Block 868. Block 868 samples either or both the I (In-phase) and Q (quadrature) linearly modulated signals at the outputs of the linear decoder block 850. The signal samples are written directly to RAM with the aid of an address counter and in response to a control signal from the processor 906.

A voice can be sent as an SSB signal occupying a single voice bandwidth on the channel, or equivalently on either of the I or Q channels as was described earlier. Each of the I and Q signals simultaneously occupy the same RF bandwidth as two analog-single sidebands (SSB). Voice bandwidths are on the order of 2.8 KHz, so a typical signal sampling rate of about 6.4 KHz each is required of the analog-to-digital converter if analog-SSB is recovered from the I and Q channel information. The analog-to-digital converter samples with 8 bit precision (although as much as 10 bits is preferred). Direct memory access by the analog-to-digital converter allows the use of a processor whose speed and power are not a direct function of the channel data rate. That is, a microprocessor can be used with direct memory access, whereas, a significantly higher speed processor would be required if the analog-to-digital converted data were read to memory through the microprocessor.

The analog-to-digital converter (A/D), the dual port RAM and the address counter are grouped as block **868**. A second RAM I/O port can be serial or parallel, and operates at a 6 or 12K sample per second rate. A second RAM I/O port is provided so that the processor can extract the sampled voice or data, process the demodulation function, and expand the compressed voice or format the data. The restored voice is played back through the voice processor **914** and transducer **916**, while formatted data can be displayed on display **911**.

Again, referring to FIG. **9**, an expanded electrical block diagram is used to describe in further detail the receiver operation of the dual mode communication receiver of the present invention. The transmitted information signal, modulated in the FM modulation format, or in a linear modulation format (such as SSB), is intercepted by the antenna **802** which couples the information signal to the receiver section **902**, and in particular to the input of the radio frequency (RF) amplifier **806**. The message information is transmitted on any suitable RF channel, such as those in the VHF bands and UHF bands. The RF amplifier **806** amplifies the received information signal, such as that of a signal received on a 930 MHz paging channel frequency, coupling the amplified information signal to the input of the first mixer **808**. The first oscillator signal, which is generated in the preferred embodiment of the present invention by a frequency synthesizer or local oscillator **810**, also couples the first mixer **808**. The first mixer **808** mixes the amplified information signal and the first oscillator signal to provide a first intermediate frequency, or IF, signal, such as a 45 MHz IF signal, which is coupled to the input of the first IF filter **812**. It will be appreciated that other IF frequencies can be utilized as well, especially when other paging channel frequencies are utilized. The output of the IF filter **812** which is the on-channel information signal, is coupled to the input of the second conversion section **814**, which will be described in further detail below. The second conversion section **814** mixes the on-channel information signal to a lower intermediate frequency, such as 455 KHz, using a second oscillator signal, which is also generated by the synthesizer **810**. The second conversion section **814** amplifies the resultant intermediate frequency signal, to provide a second IF signal which is suitable to be coupled to either the FM demodulator section **908** or to the linear output section **824**.

Receiver section **804** operates in a manner similar to a conventional FM receiver, however, unlike a convention FM receiver, the receiver section **804** of the present invention also includes an automatic frequency control section **816** which is coupled to the second conversion section **814**, and which appropriately samples the second IF signal to provide a frequency correction signal which is coupled to the frequency synthesizer **810** to maintain the receiver tuning to the assigned channel. The maintenance of receiver tuning is especially important for the proper reception of QAM (that is, I and Q components) and/or SSB information which is transmitted in the linear modulation format. The use of a frequency synthesizer to generate the first and second oscillator frequencies enables the operation selection of the receiver on multiple operating frequencies, selected such as by code memory programming and/or by parameters received over the air, as for example, in the FLEX™ protocol. It will be appreciated that other oscillator circuits, such as fixed frequency oscillator circuits which can be adjusted by a frequency correction signal from the automatic frequency control section **816**, can be utilized as well.

An automatic gain control **820** is also coupled to the second conversion section **814** of the dual mode receiver of the present invention. The automatic gain control **820** estimates the energy of samples of the second IF signal and provides a gain correction signal which is coupled to the RF amplifier **806** to maintain a predetermined gain for the RF amplifier **806**. The gain correction signal also couples the second conversion section **814** to maintain a predetermined gain for the second conversion section **814**. The maintenance of the gain of the RF amplifier **806** and the second conversion section **814** is required for proper reception of the high speed data information transmitted in the linear modulation format, and further distinguishes the dual mode receiver of the present invention from a conventional FM receiver.

When the message information or control data is transmitted in the FM modulation format, the second IF signal is coupled to the FM demodulator section **908**, as will be explained in detail below. The FM demodulator section **908** demodulates the second IF signal in a manner well known to one of skill in the art, to provide a recovered data signal, which is a stream of binary information corresponding to the received address and message information transmitted in the FM modulation format. The recovered data signal coupled to the input of a microcomputer **906**, which function as a decoder and controller, through an input of input/output port, or I/O port **828**. The microcomputer **906** provide complete operational control of the communication receiver **900**, providing such functions as decoding, message storage and retrieval, display control, and alerting, just to name a few. The device **906** is preferably a single chip microcomputer such as the MC68HC05 microcomputer manufactured by Motorola, and includes CPU **840** for operational control. An internal bus **830** connects each of the operational elements of the device **906**. I/O port **828** (shown split in FIG. 9) provides a plurality of control and data lines providing communications to device **906** from external circuits, such as the battery saver switch **904**, audio processor **914**, a display **911**, and digital storage **868**. A timing means, such as timer **834** is used to generate the timing signals required for the operation of the communication receiver, such as for battery saver timing, alert timing, and message storage and display timing. Oscillator **832** provides the clock for operation of CPU **840**, and provides the reference clock for timer **834**. RAM **838** is used to store information utilized in executing the various firmware routines controlling the operation of the communication receiver **900**, and can also be used to store short messages, such as numeric messages. ROM **836** contains the firmware routines used to control the device **906** operation, including such routines as required for decoding the recovered data signal, battery saver control, message storage and retrieval in the digital storage section **868**, and general control of the pager operation and message presentation. An alert generator **842** provides an alerting signal in response to decoding the FM modulated signaling information. A code memory **910** (not shown) couples the microcomputer **906** through the I/O port **828**. The code memory is preferably an EEPROM (electrically erasable programmable read only memory) which stores one or more predetermined addresses to which communication receiver **900** is responsive.

When the FM modulated signaling information is received, it is decoded by the device **906**, functioning as a decoder in a manner well known to one skilled in the art. When the information in the recovered data signal matches any of the stored predetermined addresses, the subsequently received information is decoded to determine if additional information is directed to the receiver which is modulated in the FM modulation format, or if the additional information is modulated in the linear modulation format. When the additional information is transmitted in the FM modulation format, the recovered message information is received and stored in the microcomputer RAM **838**, or in the digital storage section **868**, as will be explained further below, and an alerting signal is generated to alert generator **842**. The alerting signal is coupled to the audio processing circuit **914** which drives transducer **916**, delivering an audible alert. Other forms of sensible alerting, such as tactile or vibrating alert, can also be provided to alert the user as well.

When additional information is to be transmitted in the linear modulation format (such as SSB or "I and Q"), the microcomputer **906** decodes pointer information. The pointer information includes information indicating to the receiver on what combination of sidebands (or on what combination of I and Q components) within the channel bandwidth that the additional information is to be transmitted. The device **906** maintains the operation of monitoring and decoding information transmitted in the FM modulation format, until the end of the current batch, at which time the supply of power is suspended to the receiver until the next assigned batch, or until the batch identified by the pointer is reached, during which high speed data is transmitted. The device **906**, through I/O port **828** generates a battery saving control signal which couples to battery saver switch **904** to suspend the supply of power to the FM demodulator **908**, and to supply power to linear output section **824**, the linear demodulator **850**, and the digital storage section **868**, as will be described below.

The second IF output signal, which now carries the SSB (or "I and Q") information is coupled to the linear output section **824**. The output of the linear output section **824** is coupled to the quadrature detector **850**, specifically to the input of the third mixer **852**. A third local oscillator also couples to the third mixer **852**, which is preferably in the range of frequencies from 35–150 kHz, although it will be appreciated that other frequencies may be utilized as well. The signal from the linear output section **824** is mixed with the third local oscillator signal **854**, producing a third IF signal at the output of the third mixer **852**, which is coupled to a third IF amplifier **856**. The third IF amplifier is a low gain amplifier which buffers the output signal from the input signal. The third output signal is coupled to an I channel mixer **858** and a Q channel mixer **860**. The I/Q oscillator **862** provides quadrature oscillator signals at the third IF frequency which are mixed with the third output signals in the I channel mixer **858** and the Q channel mixer **860**, to provide baseband I channel signals and Q channel signals at the mixer outputs. The baseband I channel signal is coupled to a low pass filter **864**, and the baseband Q channel signal is coupled to a low pass filter **866**, to provide a pair of baseband audio signals which represent the compressed and companded voice signals.

The audio signals are coupled to the digital storage section **868**, in particular to the inputs of an analog to digital converter **870**. The A/D converter **870** samples the signals at a rate at least twice the highest frequency component at the output of **864** and **866**. The sampling rate is preferably 6.4 kilohertz per I and Q channel. It will be appreciated, that the data sampling rate indicated is for example only, and other sampling rates may be used depending upon the bandwidth of the audio message received.

During the batch when the high speed data is transmitted, the microprocessor **906** provides a count enabling signal which is coupled to the address counter **872**, the A/D converter **870** is also enable to allow sampling of the information symbol pairs. The A/D converter **870** generates high speed sample clock signals which are used to clock the address counter **872** which in turn sequentially generates addresses for loading the sampled voice signals into a dual port random access memory **874** through data lines going from the converter **870** to the RAM **874**. The voice signals which have been loaded at high speed into the dual port RAM **874** in real time, are processed by the microcomputer **906** after all voice signals have been received, thereby producing a significant reduction in the energy consumed by not requiring the microcomputer **906** to process the information in real time. The microcomputer **906** accesses the stored signals through data lines and address lines, and in the preferred embodiment of the present invention, processes the information symbol pairs to generate either ASCII encoded information in the case of alphanumeric data having been transmitted, or digitized sampled data in the case voice was transmitted. The digitized voice samples can alternatively stored in other formats such as BCD, CVSD, or LPC based forms and other types as required. In the case of time compressed voice signals, the I and Q components sampled by ADC converter **870** are further processed by CPU **840** via dual port RAM **874** and I/O **828** to (1) amplitude expand the audio signal and (2) time-expand the signal as was described in the similar operation of the receivers of FIGS. **7** and **8**. The voice is then stored again in RAM **874**. The ASCII encoded or voice data is stored in the dual port RAM until the information is requested for presentation by the communication receiver user. The stored ASCII encoded data is recovered by the user using switches (not shown) to select and read the stored messages. When the stored ASCII encoded message is to be read, the user selects the message to be read and actuates a read switch which enable microcomputer **906** to recover the data, and to present the recovered data to a display **911**, such as a liquid crystal display. When a voice message is to be read, the user selects the message to be read and actuates a read switch which enables the microcomputer **906** to recover the data from the dual port RAM, and to present the recovered data to the audio processor **914** which converts the digital voice information into an analog voice signal which is coupled to a speaker **916** for presentation of the voice message to the user. The microcomputer **906** can also generate a frequency selection signal which is coupled to frequency synthesizer **810** to enable the selection of different frequencies as previously described.

Referring to FIG. **10**, a timing diagram is shown which illustrates features of the FLEX™ coding format on outbound signaling utilized by the radio communication system **100** of FIG. **1**, and which includes details of a control frame **330**, in accordance with the preferred embodiment of the present invention. Control frames are also classified as digital frames. The signaling protocol is subdivided into protocol divisions, which are an hour **310**, a cycle **320**, frames **330**, **430** a block **340**, and a word **350**. Up to fifteen 4 minute uniquely identified cycles are transmitted in each hour **310**. Normally, all fifteen cycles **320** are transmitted each hour. Up to one hundred twenty eight 1.875 second uniquely identified frames including digital frames **330** and analog frames **430** are transmitted in each of the cycles **320**. Normally, all one hundred twenty eight frames are transmitted. One synchronization and Frame Information signal **331** lasting one hundred fifteen milliseconds and 11 one hundred sixty millisecond uniquely identified blocks **340** are transmitted in each of the control frames **330**. Bit rates of

3200 bits per second (bps) or 6400 bps are preferably used during each control frame **330**. The bit rate during each control frame **330** is communicated to the selective call radios **106** during the synchronization signal **331**. When the bit rate is 3200 bps, 16 uniquely identified 32 bit words are included in each block **340**, as shown in FIG. **10**. When the bit rate is 6400 bps 32 uniquely identified 32 bit words are included in each block **340** (not shown). In each word, at least 11 bits are used for error detection and correction, and 21 bits or less are used for information, in a manner well known to one of ordinary skill in the art. The bits and words **350** in each block **340** are transmitted in an interleaved fashion using techniques well known to one of ordinary skill in the art to improve the error correction capability of the protocol.

Information is included in each control frame **330** in information fields, comprising Frame structure information in a block information field (BI) **332**, one or more selective call addresses in an address field (AF) **333**, and one or more vectors in a vector field (VF) **334**. The vector field **334** starts at a vector boundary **334**. Each vector in the vector field **334** corresponds to one of the addresses in the address field **333**. The boundaries of the information fields **332**, **333**, **334** are defined by block information field **332**. Information fields **332**, **333**, **334** are variable, depending on factors such as the type of system information included in the sync and frame information field **331** and the number of addresses included in the address field **333**, and the number and type of vectors included in the vector field **334**.

Referring to FIG. **11**, a timing diagram is shown which illustrates features of the transmission format of the outbound signaling protocol utilized by the radio communication system of FIG. **1**, and which includes details of a voice frame **430**, in accordance with the preferred embodiment of the present invention. Voice frames are also classified herein as analog frames. The durations of the protocol divisions hour **310**, cycle **320**, and frame **330**, **430** are identical to those described with respect to a control frame in FIG. **10**. Each analog frame **430** has a header portion **435** and an analog portion **440**. The information in the synchronization and frame information signal **331** is the same as the synchronization signal **331** in a control frame **330**. As described above, the header portion **435** is frequency modulated and the analog portion **440** of the frame **430**, is amplitude modulated. A transition portion **444** exists between the header portion **435** and analog portion **440**. In accordance with the preferred embodiment of the present invention, the transition portion includes amplitude modulated pilot subcarriers for up to three subchannels **441**, **442**, **443**. The analog portion **440** illustrates the three subchannels **441**, **442**, **443** which are transmitted simultaneously, and each subchannel includes an upper sideband signal **401** and a lower sideband signal **402** (or alternatively, an in-phase and a quadrature signal). In the example illustrated in FIG. **11**, the upper sideband signal **401** includes one message fragment **415**, which is a first fragment of a first analog message. Included in the lower sideband **402** are four quality assessment signals **420**, **422**, **424**, **426**, four message segments **410**, **412**, **416**, **418**, and one segment **414** (unused in this example). The two segments **410**, **412** are segments of a second fragment of the first analog message. The two segments **416**, **418** are segments of a first fragment of a second analog message. The first and second analog messages are compressed voice signals which have been fragmented for inclusion in the first subchannel **441** of frame one **430** of cycle 2 of **320**. The second fragment of the first message and the first fragment of the second message are each split to include a quality assessment signal **420**, **426**, which are repeated at predetermined positions in the lower sideband **402** of each of the three subchannels **441**, **442**, **443**. The smallest segment of message included in an analog frame is defined as a voice increment **450**, of which **88** are uniquely identified in each analog portion **440** of an analog frame **430**. The quality assessments signals are preferably transmitted as unmodulated subcarrier pilot signals, are preferably one voice increment in duration, and preferably have a separation of no more than 420 milliseconds within an analog portion of a frame. It will be appreciated that more than one message fragment could occur between two quality assessment signals, and that message fragments are typically of varying integral lengths of voice increments.

Referring to FIG. **12**, a timing diagram illustrating a control frame **330** and two analog frames of the outbound signaling protocol utilized by the radio communication system of FIG. **1** is shown, in accordance with the preferred embodiment of the present invention. The diagram of FIG. **12** shows an example of a frame zero (FIG. **10**) which is a control frame **330**. Four addresses **510**, **511**, **512**, **513** and four vectors **520**, **521**, **522**, **523** are illustrated. Two addresses **510**, **511** include one selective call radio **106** address, while the other two addresses **512**, **513** are for a second and third selective call radio **106**. Each address **510**, **511**, **512**, **513** is uniquely associated with one of the vectors **520**, **521**, **522**, and **523** by inclusion of a pointer within each address which indicates the protocol position of (i.e., where the vector starts and how long it is) the associated vector.

In the example shown in FIG. **12**, vectors **520**, **521**, **522**, **523** are also uniquely associated with a message portion in one of the subchannels. Specifically, vector **520** can point to an upper sideband of subchannel **441** (see FIG. **11**) and vector **522** can point to a lower sideband of subchannel **441**. Similarly, vector **521** can point to both sidebands of subchannel **442**. That is, in the case of subchannel **441**, the example can show that two different message portions are carried by the upper and lower sidebands. In the case of subchannel **442**, two halves of one message portion are carried by the upper and lower sidebands respectively. Thus, the vectors preferably include information therein to indicate which subchannel (i.e., which radio frequency) the receiver should look for a message, and also information to indicate whether two separate messages are to be recovered from the subchannel, or whether first and second halves of a single message are to be recovered.

One use for the embodiment where two different messages are simultaneously transmitted over upper and lower sidebands (or I and Q channels), respectively, is where one message is a direct voice paging message, and the other is a voice mailbox message, which is to be stored in the pager.

In accordance with the preferred embodiment of the present invention the vector position is provided by identifying the number of words **350** after the vector boundary **335** at which the vector starts, and the length of the vector, in words. It will be appreciated that the relative positions of the addresses and vectors are independent for each other. The relationships are illustrated by the arrows. Each vector **520**, **521**, **522**, **523** is uniquely associated with a message fragment **550**, **551**, **552**, **553** by inclusion of a pointer within each vector which indicates the protocol position of (i.e., where the fragment starts and how long it is) the associated vector. In accordance with the preferred embodiment of the present invention the message fragment position is provided by identifying the frame **430** number (from 1 to 127), the subchannel **441**, **442**, **443** number (from one to three), the sideband **401**, **402**, (or I or Q) and the voice increment **450** where the message fragment starts, and the length of the message fragment, in terms of voice increments **450**. For example, vector three **522** includes information which indicates that message two, fragment one **552**, which is intended for selective call transceiver **106** having selective call address **512**, is located starting at voice increment forty six **450** (the voice increments **450** are not identified in FIG. **12**) of frame one **560**, and vector thirteen **523** includes information which indicates that message nine fragment one **553**, which is intended for selective call transceiver **106** having selective call address **513**, is located starting at voice increment zero **450** (the voice increments **450** are not shown in FIG. **12**) of frame five **561**.

It will be appreciated that, while voice signals are described in accordance with the preferred embodiment of the present invention, other analog signals, such as modem signals or dual tone multifrequency (DTMF) signals, can alternatively be accommodated by the present invention. It should also be appreciated that the block information used in the frame structure previously described can be used to implement further enhancements that would allow for greater overall throughput in a communication system and allow for additional features. For instance, a message sent to a portable voice unit can request that an acknowledgment signal sent back to the system include information that would identify the transmitter it was receiving its messages from. Thus, frequency reuse in a simulcast system can be achieved in this way by transmitting messages to the given portable voice unit using the one transmitter required to reach the portable voice unit. Additionally, once the system knows the location of the portable voice unit, implementing target messaging logically follows.

In another aspect of the present invention, the time-scaling technique, previously described as WSOLA has some existing disadvantages when used in conjunction with the present invention. Thus, a technique was developed that modifies WSOLA to become speaker dependent and appropriately named "WSOLA-SD". To further understand our modification of WSOLA to form WSOLA-SD, a brief description of WSOLA follows.

A technique called Waveform similarity based Overlap-Add technique (WSOLA) can achieve high-quality time-scale modification compared to other techniques and is also much simpler than other methods. When used to speed up or slow down speech, the quality of speech is not very good even with the WSOLA technique. The reconstructed speech contains a lot of artifacts like echoes, metallic sounds and reverberations in the background. This aspect of the present invention describes several enhancements to overcome this problem and minimize the artifacts present. Many parameters in the WSOLA algorithm have to be optimized to achieve the best quality possible for a given speaker and required compression/expansion or time-scaling factor. This aspect of the invention deals with determining those parameters and how to incorporate them in compression/expansion or time-scaling of speech signals with improvement in the quality of the recovered speech or voice signal.

The WSOLA Algorithm: Let $x(n)$ be the input speech signal to be modified, $y(n)$ the time-scale modified signal and α be the time-scaling parameter. If α is less than 1 then the speech signal is expanded in time. If α is greater than 1 then the speech signal is compressed in time.

Referring to FIGS. **13–17**, timing diagrams for several iterations of the WSOLA time-scaling (compression) method is shown for comparison to the preferred method of WSOLA-SD of the present invention. Assuming that the input speech signals are appropriately digitized and stored, FIG. **13** illustrates the first iteration of the WSOLA method on an uncompressed speech input signal. The WSOLA method requires a time scale factor of α (which we assume is equal to 2 for this example, where if $\alpha>1$ we have compression and if $\alpha<1$ we have expansion) and an arbitrary analysis segment size (Ss) which is independent of the input speech characteristics, and in particular, independent of pitch. An overlap segment size So is computed as 0.5*Ss and is fixed in WSOLA. The first Ss samples are copied directly to the output as shown in FIG. **14**. Let the index of the last sample in the output be L_{f1}. An overlap index O_1 is determined as Ss/2 samples from the end of the last available sample in the output. Now the samples which would be overlap added are between O_1 and L_{f1}. Search index (S_1) is determined as α^*O_1. After an initial portion of the input signal is copied into the output, a determination is made of the moving window of samples from the input. The window is determined around the search index S_1. Let the beginning of the window be S_i-L_{offset} and the end be S_i+H_{offset}. In the first iteration, i=1. Within the window, the best correlating So samples are determined using a Normalized Cross-Correlation equation given by:

$$R(k)=\frac{\sum_{j=0}^{j=S_0} x(S_i+k+j)y(O_i+j)}{\left[\sum_{j=0}^{j=S_0} x^2(S_i+k+j)\sum_{j=0}^{j=S_0} y^2(O_i+j)\right]^{1/2}}$$

where

$$k=S_i-L_{offset},\ S_i+H_{offset}$$

The lag k=m for which the normalized R(k) is maximum is determined. The best index Bi is given by Si+m. Note that other schemes like Average Magnitude Difference Function (AMDF) and other correlation functions can be used to find the best matching waveform. The So samples beginning at B1 are then multiplied by an increasing ramp function (although other weighting functions could be used) and added to the last So samples in the output. Prior to the addition, the So samples in the output are multiplied by a decreasing ramp function (although other weighting functions could be used here as well). The resulting samples of the addition will replace the last So samples in the input. Finally, the next So samples which immediately follow the prior best matching So samples are then copied to the end of the output for use in the next iteration. This would be the end of the first iteration in WSOLA.

Referring to FIGS. **15** and **16** for the next iteration, we need to compute a new overlap index O_2, similarly to O_1. Likewise, a new search index S_2 and corresponding search window is determined as was done in the previous iteration. Once again, within the search window, the best correlating So samples are determined using the cross-correlation equation previously described above, where the beginning of the best samples determined is B_2. The So samples beginning at B2 are then multiplied by an increasing ramp function and added to the last So samples in the output. Prior to the addition, the So samples in the output are multiplied by a decreasing ramp function. The resulting samples of the addition will replace the last So samples in the input. Finally, the next So samples which immediately follow the prior best matching So samples are then copied to the end of the output for use in the next iteration, where future i^{th} iterations would have an overlap index O_i, a Search index S_i, last sample in output L_{fi}, and a best index B_i.

FIG. **17** shows the resultant output from the previous two iterations described with reference to FIGS. **13–16**. Once should note that there is no overlap in the resultant output signal between the two iterations. If the method were to continue in a similar fashion, the WSOLA method would time scale (compress) the entire speech signal, but there would never be any overlap between the results of each of the iterations. WSOLA time-scale expansion is done in a similar fashion.

Several drawbacks or disadvantages of WSOLA with respect to the preferred method of the present invention (WSOLA-SD) become apparent. These drawbacks should be kept in mind as you follow the next examples of the WSOLA-SD method shown in FIGS. **18–23**. A primary drawback of WSOLA includes the inability to obtain the optimum quality of time scaled speech because a fixed analysis segment size (Ss) is used for all input speech irrespective of the pitch characteristics. For instance, if the Ss was too large for the input speech signal, the resultant speech upon expansion would include echoes and reverberations. Further, if the Ss is too small for the input speech signal, then the resultant speech upon expansion would sound raspy.

A second significant drawback of WSOLA results when compression rates (α) are greater than 2. In such instances, the separation of the moving window between iterations may cause the method to skip significant input speech components, thereby seriously affecting the intelligibility of the resultant output speech. Increasing the size of the moving windows to compensate for the non-overlapping search windows during iterations causes further skipping of some input speech as a result of the cross-correlation function and further causes variable time-scaling that noticeably affects the resultant output speech.

A third drawback of the WSOLA method involves its failure to provide a designer or user the flexibility (for a given time-scaling factor (α)) with respect to quality of speech and complexity of computation for a given system having given restraints. This is particularly apparent because the degree of overlap (f) is fixed at 0.5 in the WSOLA method. Thus, in an application that requires high quality speech reproduction, assuming adequate processing power and memory, the WSOLA-SD method of the present invention can use a higher degree of overlap at the expense of added computational complexity to provide higher quality speech reproduction. On the other hand, in an application that is limited by processing power, memory or other constraints, the degree of overlap can be lowered in WSOLA-SD so that the quality of speech is sacrificed only to the extent desired, taking into account the particular application constraints at hand.

FIG. **25** illustrates an overall block diagram of WSOLA-SD method. In this block diagram Ss, f and α are computed depending on whether we are compressing or expanding speech. This WSOLA-SD algorithm provides great improvement in the quality of reconstructed speech over WSOLA alone. The WSOLA-SD method is speaker dependent, particularly to the pitch of a particular speaker. Thus, a pitch determination **12** is done before an analysis segment sized is determined (**14**). For a given f and α (which can be modified dependent upon the pitch determination **12**, providing a modified alpha (**16**)), WSOLA-SD time scales (**18**) the speech. The time-scaling can either be expansion or compression of the input signal. Alternatively, frequency-scaled signal can be obtained by interpolating the time-scaled signal by a factor of α if $\alpha>1$ or by decimating the time-scaled signal by a factor of $1/\alpha$ if $\alpha<1$. Interpolation and decimation are well known techniques in digital signal processing as described in Discrete Time Signal Processing by Oppenheim & Schaefer. For example, assuming 2 seconds worth of an input speech is sampled at 8 kHz, where the signal has significant frequency components between 0 and 4000 Hz. Assuming the input speech signal is time-scale compressed by a factor of 2. The resultant signal would have a length of 1 second, but would still have significant frequency components between 0 and 4000 Hertz. The signal is interpolated (See Oppenheim & Schaefer) by a factor of $\alpha=2$. This would result in a signal which is 2 seconds long, but with frequency component between 0 and 2000 Hertz. Returning to the time scale domain can be achieved by decimating the frequency compressed signal by a factor of $\alpha=2$ to obtain the original time scaled speech (frequency components between 0–4000 Hertz) without any loss of information content.

Referring to FIGS. **18–22**, timing diagrams for several iterations of the WSOLA-SD time-scaling (compression) method is shown in accordance with the present invention. Assuming that the input speech signals are appropriately digitized and stored, FIG. **18** illustrates the first iteration of the WSOLA-SD method on an uncompressed speech input signal. The WSOLA-SD method also requires the determination of an approximate pitch period of the voiced portions of the input speech signal. A brief description of the pitch determination and how the segment size is obtained from it is given below.

1) Frame input speech into 20 ms blocks.
2) Compute energy in each block.
3) Compute average energy per block.
4) Determine energy threshold to detect voiced speech as a function of the average energy per block.
5) Using the energy threshold determine contiguous blocks of voiced speech of a length of at least 5 blocks.
6) On each block of the contiguous voice speech found in step 5, do a pitch analysis. This could be done using a variety of methods including Modified Auto correlation method, AMDF or Clipped auto correlation method.
7) The pitch values are smoothened using a median filter to eliminate errors in the estimation.
8) Average all the smoothened pitch values to obtain an approximate estimate of the speaker's pitch
9) Thus, the Segment size Ss computation is given below.

If pitch P greater than 60 samples Ss=2*Pitch
If pitch P is between 40 and 60 samples Ss=120
If P less than 40 samples Ss=100

A sampling rate of 8 Khz is assumed in all cases above. A critical factor that provides WSOLA-SD with the advantages that overcomes some of the drawbacks previously described above in the description of WSOLA is the degree of overlap f. If the degree of overlap f in WSOLA-SD is greater than 0.5, then this provides higher quality at the expense of more complexity. If the degree of overlap f in WSOLA-SD is less than 0.5, then this reduces complexity of the algorithm at the expense of quality. Thus, the user has more flexibility and control in design and use of their particular application.

Again, referring to FIGS. **18–23**, the WSOLA-SD method requires a time scale factor of α (which we assume is equal to 2 for this example, where if $\alpha>1$ we have compression and if $\alpha<1$ we have expansion) and an analysis segment size (Ss) which is optimized to the input speech characteristics, namely the pitch of the speaker. An overlap segment size So is computed as f*Ss and is fixed in WSOLA-SD for a given pitch period and f. In the example shown, f is greater than 0.5, to show higher quality resultant output speech. The first

Ss samples are copied directly to the output. Let the index of the last sample be I_{f1}. An overlap index O_1 is determined as So samples from the end of the last available sample in the output. Now the samples which would be overlap added are between O_1 and I_{f1} as shown in FIG. 19. The first search index (S_1) is determined as $\alpha*O_1$ as seen in FIG. 18. After an initial portion of the input signal is copied into the output, a determination is made as to the location of the moving window of samples from the input speech signal. The window is determined around or about the search index S_1. Within the window, the best correlating So samples are determined using the cross-correlation equation previously described above, where the beginning of the best samples determined is B_1. The So samples beginning at B1 are then multiplied by an increasing ramp function (although other weighting functions could be used) and added to the last So samples in the output. Prior to the addition, the So samples in the output are multiplied by a decreasing ramp function. The resulting samples of the addition will replace the last So samples in the input. Finally, the next Ss-So samples which immediately follow the prior best matching So samples are then copied to the end of the output for use in the next iteration. This would be the end of the first iteration in WSOLA-SD.

Referring to FIGS. 20 and 21 for the next iteration, we need to compute a new overlap index O_2, similarly to O_1. Likewise, a new search index S_2 and corresponding search window is determined as done in the previous iteration. Once again, within the search window, the best correlating So samples are determined using the cross-correlation equation previously described above, where the beginning of the best samples determined is B_2. The So samples beginning at B2 are then multiplied by an increasing ramp function and added to the last So samples in the output. Prior to the addition, the So samples in the output are multiplied by a decreasing ramp function. The resulting samples of the addition will replace the last So samples in the input. Finally, the next Ss-So samples which immediately follow the prior best matching So samples are then copied to the end of the output for use in the next iteration.

FIG. 22 shows a resultant output signal from two iterations using the WSOLA-SD method. Note that there is a region of overlap (Ss-So) in the resultant output signal which insures increased intelligibility and prevents the method from skipping critical input speech components as compared to the WSOLA method.

Referring to FIGS. 23 and 24, an i^{th} iteration of an example input timing diagram and output timing diagram for time-scale expansion using the WSOLA-SD method is shown in accordance with the present invention. The method for expansion essentially functions similarly to the examples shown in FIGS. 18–22 except that O_i, the overlap index, moves faster than the S_i, the Search index. To be exact, O_i moves α times faster than S_i during expansion. The analysis segment size Ss is dependent on the pitch period of the input speech. The degree of overlap can range from 0 to 1, but 0.7 is used for this example in FIGS. 23 and 24. The time scaling factor α, in this instance, will be the inverse of the expansion rate. Assuming the expansion rate was 2, then the time scaling factor α=0.5. The overlap segment size So would equal f*Ss or the degree of overlap times the analysis segment size. Thus, after several iterations of overlap adding and using an increasing ramp function on each best matching input segment and using a decreasing ramp function on each output overlap segment, prior to the addition, the input speech signal is expanded as the output speech signal that maintains all the advantages of WSOLA-SD as previously described.

Further improvement is obtained by dynamically adapting the segment size Ss in the WSOLA-SD algorithm with the pitch of the segment at that instant. This is done by a modification of the scheme explained previously. If we use a short segment size of Ss=100(sampling rate 8 Khz is assumed) for unvoiced speech sounds their quality is improved and for voiced speech the segment size will be Ss=2*Pitch. Also a few changes are necessary to determine whether the speech segment is voiced or unvoiced. The method with these changes is described below.

1) Frame input speech into 20 ms blocks.
2) Compute energy in each block.
3) Compute number of zero-crossings in each block.
4) Compute average energy per block.
5) Determine energy threshold to detect voiced speech as a function of the average energy per block.
5) Using the energy threshold and zero-crossing threshold determine,contiguous blocks of voiced speech of length of at least 5 blocks.
6) Do pitch analysis on all the voiced segments and determine the average pitch in each of those voiced segments. This could be done using a variety of methods including Modified Auto correlation method, AMDF or Clipped auto correlation method.
7) The segments that are not marked as Voiced speech are now marked as tentative unvoiced segments.
8) Contiguous blocks of at least 5 frames in the 'tentative unvoiced segments' are taken and pitch analysis is done. The ratio of the maximum to minimum correlation coefficient is determined. If the ratio is large then the segment is classified as Unvoiced or if it is small these segments are marked as voiced and average pitch of those segments are determined along with the start and ending of the speech segment.
9) Segment size Ss for each of these classified speech segments are determined as follows.

If Voiced Ss=2*Pitch

If Unvoiced Ss=100 (Sampling rate of 8 Khz is assumed)

10) Now WSOLA-SD method of time-scaling is done, but with a varying segment size. Here the position of the input speech segment used in the processing at each time instant is determined. Depending on its position, the segment sizes Ss already determined is used in the processing. Using this technique results in a higher quality time-scaled speech signal.

If WSOLA-SD is used to do both compression and then a subsequent expansion on the same speech input signal as in the case of our communication system, the quality of the reconstructed speech signal can be further improved for a given average time-scale factors using several techniques.

From perceptual tests, it can be seen that a speech signal which has a higher fundamental frequency (lower pitch period) can be compressed more for a given speech quality as compared to a speech signal which has a lower fundamental frequency (higher pitch period). For instance, children and female speakers will on average have a higher fundamental frequency. Thus, their speech can be compressed/expanded by 10% more without noticeably affecting the quality of their speech. Whereas male speakers who have speech on average with a lower fundamental frequency, can have their speech compressed/expanded by 10% less. Thus, in a typical communication system having roughly equal number of speakers having higher and lower fundamental frequencies, an overall improved quality in the reproduction of speech is obtained with the same compression/expansion (time-scaling) factor as before.

Another characteristic of expansion and compression using this technique leads to further enhancements. For instance, it was noticed that most of the artifacts in the speech are produced during the time-scale expansion of the speech signal. The more the speech signal is expanded the more the artifacts. It was also observed that if the speech signal is played back a little faster (less than 10%) than the original speech, the change in speed is hardly noticeable, but with a noticeable reduction in artifacts. This property helps expand the speech signal with a smaller expansion factor and thus reduce the artifacts and improve its quality. For example, if the input speech is compressed by a time-scaling factor of 3, then during expansion it would be expanded by a factor of 2.7, which means that the speech will be played faster by 10%. Since this change in speech rate will not be noticeable and reduces artifacts, it should be implemented in the method of the present invention in applications where the accuracy of the speech is not absolutely critical.

What is claimed is:

1. A method for compressing a plurality of voice signals within a voice communication resource having a given bandwidth within a voice communication system, comprising the steps of:

(a) subchanneling the voice communication resource into a plurality of subchannels and simultaneously placing a pair of the plurality of voice signals on a subchannel;

(b) modulating the pair of the plurality of voice signals about a pilot signal within the subchannel within the voice communication resource using single sideband modulation; and

(c) compressing the time of each of the voice signals within the plurality of subchannels, wherein the result of steps (a), (b), and (c) provides a compressed voice signal.

2. The method of claim 1, wherein the step of subchanneling further comprises the step of using quadrature amplitude modulation.

3. The method of claim 1, wherein the step of compressing the time of each of the voice signals further comprises the step using time-scale compression on the voice signals.

4. The method of claim 1, wherein the step of compressing the time of each of the voice signals further comprises the step of using Waveform Similarity based Overlap-Add (WSOLA) time compression on the voice signals.

5. The method of claim 1, wherein the step of compressing the time of each of the voice signals further comprises the speaker dependent steps of identifying pitch periods within each of the voice signals and transmitting data from one pitch period to alter a time-scaling factor.

6. The method of claim 1, wherein the step of compressing the time of each of the voice signals comprises the step of using a speaker dependent modification of the Waveform Similarity based Overlap-Add (WSOLA) time compression technique on the voice signals.

7. A method for compressing a plurality of voice signals within a voice communication resource within a voice communication system, comprising the steps of:

(a) subchanneling the voice communication resource into a plurality of subchannels and simultaneously placing a pair of the plurality of voice signals on a subchannel;

(b) modulating the pair of the plurality of voice signals about a pilot signal within the subchannel within the voice communication resource using single sideband modulation; and

(c) compressing the time of each of the voice signals within the plurality of subchannels, wherein the result of steps (a), (b), and (c) provides a compressed voice signal for transmission via a transmitter.

8. The method of claim 7, wherein the method further comprises the step at the transmitter of transmitting the compressed voice signal to a plurality of selective call receivers.

9. The method of claim 7, wherein the method further comprises the step of receiving the compressed voice signal and demodulating the compressed bandwidth signals at one of the the plurality of selective call receivers.

10. A communication system using voice compression having at least one transmitter base station and a plurality of selective call receivers, comprising:

at the transmitter base station:

an input device for receiving an audio signal;

a processing device for compressing the audio signal using time-scale compression and a single side band modulation technique to provide a processed signal; and

a pilot carrier signal generator that generates a pilot carrier for a pair of single side band signals which includes the processed signal, wherein the pilot carrier serves as an amplitude and phase reference for distortion that occurs as a result of channel aberrations; and

a transmitter for transmitting the processed signal;

at each of the plurality of selective call receivers:

a selective call receiver for receiving the transmitted processed signal;

a receiver circuit for detecting, filtering and responding to the amplitude and phase reference generated by the pilot carrier signal generator;

a processing device for demodulating the received processed signal using single side band demodulation and time-scale expansion to provide a reconstructed signal; and

an amplifier for amplifying the reconstructed signal into an reconstructed audio signal.

11. The communication system of claim 10, wherein the single sideband modulation technique provides for the transmission of a single message split between an upper sideband and a lower sideband.

12. The communication system of claim 10, wherein the single sideband modulation technique provides for the transmission of a single message repeated on an upper sideband and lower sideband.

13. A selective call receiver for receiving compressed voice signals, comprising:

a selective call receiver for receiving a transmitted processed signal that includes compressed voice signals that have been compressed using time-scale compression;

a processing device for demodulating the received processed signal, wherein said processing device demodulates both and upper and a lower sideband of a subchannel, the upper and lower sidebands having independent information therein, and wherein said processing device uses time-scale expansion to provide a reconstructed signal;

a receiver circuit for detecting, filtering and responding to the amplitude and phase reference generated by a pilot carrier signal generator in a transmitter at a base station; and

an amplifier for amplifying the reconstructed signal into an reconstructed audio signal.

14. A selective call paging base station for transmitting selective call signals on a communication resource having a predetermined bandwidth, comprising:

an input device for receiving a plurality of audio signals;

a means for subchannelizing the communication resource into a predetermined number of subchannels;

an amplitude compression and filtering module for each subchannel for compressing the amplitude of the respective audio signal and filtering the respective audio signal;

a time compression module for compressing the time of the respective audio signal for each subchannel; and

a quadrature amplitude modulation transmitter for transmitting the processed signal.

15. The selective call paging base station of claim 14, wherein the input device for receiving a plurality of audio signals, comprises a paging terminal for receiving phone messages or data messages from a computing device.

16. The selective call paging base station of claim 14, wherein the amplitude compression and filtering module comprises an anti-alias filter coupled to an analog-to-digital converter coupled to a band-pass filter coupled to an automatic gain controller and clipper circuit.

17. The selective call paging base station of claim 14, wherein the time compression module comprises a processing device for compressing the audio signal using a time-scale compression technique.

18. The selective call paging base station of claim 14, wherein the time compression module comprises a processing device for compressing the audio signal using a WSOLA time compression technique.

19. A selective call receiver unit for receiving compressed voice selective call signals, comprising:

a receiver having a analog to digital converter for providing a digitized received signal;

a digital signal processor for performing single sideband demodulation of a subchannel having a pilot carrier and independent information on an upper and a lower sideband of a subchannel, wherein the digital signal processor also performs the functions of filtering the pilot carrier, performing automatic gain control using a feedforward loop, and decompanding the digitized received signal to provide a processed signal; and

a digital to analog converter and reconstruction filter for converting the processed signal into a digitized audio signal; and

an amplifier for amplifying the digitized audio signal.

20. A communication base station, comprising:

a terminal for receiving an audio speech signal;

an analog to digital converter for converting the audio speech signal into a digitized speech signal;

a digital signal processor for processing the digitized speech signal by performing the function of splitting the digitized speech signal and at least one of the functions of bandpass filtering, automatic gain control, time scaling, companding, or buffering; and

a transmitter having at least a Hilbert transform filter coupled to a digital to analog converter coupled to a reconstruction filter coupled to a quadrature, amplitude modulator which is coupled to a radio frequency power amplifier.

* * * * *

BAR CODE LABEL

U.S. PATENT APPLICATION

SERIAL NUMBER	FILING DATE	CLASS	GROUP ART UNIT
08/395,747	02/28/95	381	2308

APPLICANT

CLIFFORD D. LEITCH, CORAL SPRINGS, FL; ROBERT J. SCHWENDEMAN, POMPANO BEACH, FL; KAZIMIERZ SIWIAK, CORAL SPRINGS, FL; WILLIAM J. KUZNICKI, CORAL SPRINGS, FL; SUNIL SATYAMURTI, DELRAY BEACH, FL.

CONTINUING DATA*******************
VERIFIED

FOREIGN/PCT APPLICATIONS**********
VERIFIED

FOREIGN FILING LICENSE GRANTED 03/20/95

STATE OR COUNTRY	SHEETS DRAWING	TOTAL CLAIMS	INDEPENDENT CLAIMS	FILING FEE RECEIVED	ATTORNEY DOCKET NO.
FL	14	24	7	$1,122.00	PT00600U

ADDRESS

JOHN H. MOORE
MOTOROLA INC
1500 GATEWAY BLVD MS 96
BOYNTON BEACH FL 33426-8292

TITLE

VOICE COMPRESSION METHOD AND APPARATUS IN A COMMUNICATION SYSTEM

This is to certify that annexed hereto is a true copy from the records of the United States Patent and Trademark Office of the application which is identified above.

By authority of the
COMMISSIONER OF PATENTS AND TRADEMARKS

Date Certifying Officer

08/395747

PATENT APPLICATION SERIAL NO. ______

U.S. DEPARTMENT OF COMMERCE
PATENT AND TRADEMARK OFFICE
FEE RECORD SHEET

PTO-1556
(5/87)

5C13045 03/09/95 08395747 18 4278 130 101 1,122.00CH PT00600U

UNITE TATES PATENT AND TRADEI. ꓤK OFFICE

Our Docket No. PT00600U
Date: February 28, 1995

Hon. Commissioner of Patents
and Trademarks
Washington, DC 20231

Sir:

Transmitted herewith for filing is the patent application of:

Inventor: Leitch, et al.
For: Voice Compression Method And Apparatus In A Communication System
Enclosed is:

_14__ sheets of formal drawings, along with 47 pages of specification and claims, and a Declaration Combined with Petition and Power of Attorney. Also enclosed is a copy of the drawing.

_____ A certified copy of a ________________ application.

__X__ List of References.

CLAIMS AS FILED

FOR	NUMBER FILED	NUMBER EXTRA	RATE	BASIC FEE $730
TOTAL CLAIMS	24 - 20 =	4	X $22	= $ 88
INDEP. CLAIMS	7 - 3 =	4	X $76	= $ 304
	TOTAL FILING FEE			$ 1,122

__X__ Please charge Deposit Account No. 13-4778 in the amount of $1,122. One original and two copies of this sheet are enclosed.

__X__ The Commissioner is hereby authorized to charge any additional fees which may be required, or credit any overpayment to Account No. 13-4778. One original and two copies of this sheet are enclosed.

Pablo Meles

Please direct all correspondence to:
John H. Moore, Esq.
Motorola, Inc.
Patent Department
1500 Gateway Boulevard
Boynton Beach, FL 33426-8292

Pablo Meles
Attorney for Applicants
Registration No. 33,739
Phone: (407) 364-2860

UNITED STATES PATENT AND TRADEMARK OFFICE

Our Docket No. PT00600U
Date: February 28, 1995

Hon. Commissioner of Patents
and Trademarks
Washington, DC 20231

Sir:

Transmitted herewith for filing is the patent application of:

Inventor: Leitch, et al.
For: Voice Compression Method And Apparatus In A Communication System
Enclosed is:

_14__ sheets of formal drawings, along with 47 pages of specification and claims, and a Declaration Combined with Petition and Power of Attorney. Also enclosed is a copy of the drawing.

_____ A certified copy of a ________________ application.

__X__ List of References.

CLAIMS AS FILED

FOR	NUMBER FILED	NUMBER EXTRA	RATE	BASIC FEE $730
TOTAL CLAIMS	24 - 20 =	4	X $22	= $ 88
INDEP. CLAIMS	7 - 3 =	4	X $76	= $ 304
TOTAL FILING FEE				$ 1,122

__X__ Please charge Deposit Account No. 13-4778 in the amount of $1,122. One original and two copies of this sheet are enclosed.

__X__ The Commissioner is hereby authorized to charge any additional fees which may be required, or credit any overpayment to Account No. 13-4778. One original and two copies of this sheet are enclosed.

Please direct all correspondence to:
John H. Moore, Esq.
Motorola, Inc.
Patent Department
1500 Gateway Boulevard
Boynton Beach, FL 33426-8292

Pablo Meles
Pablo Meles
Attorney for Applicants
Registration No. 33,739
Phone: (407) 364-2860

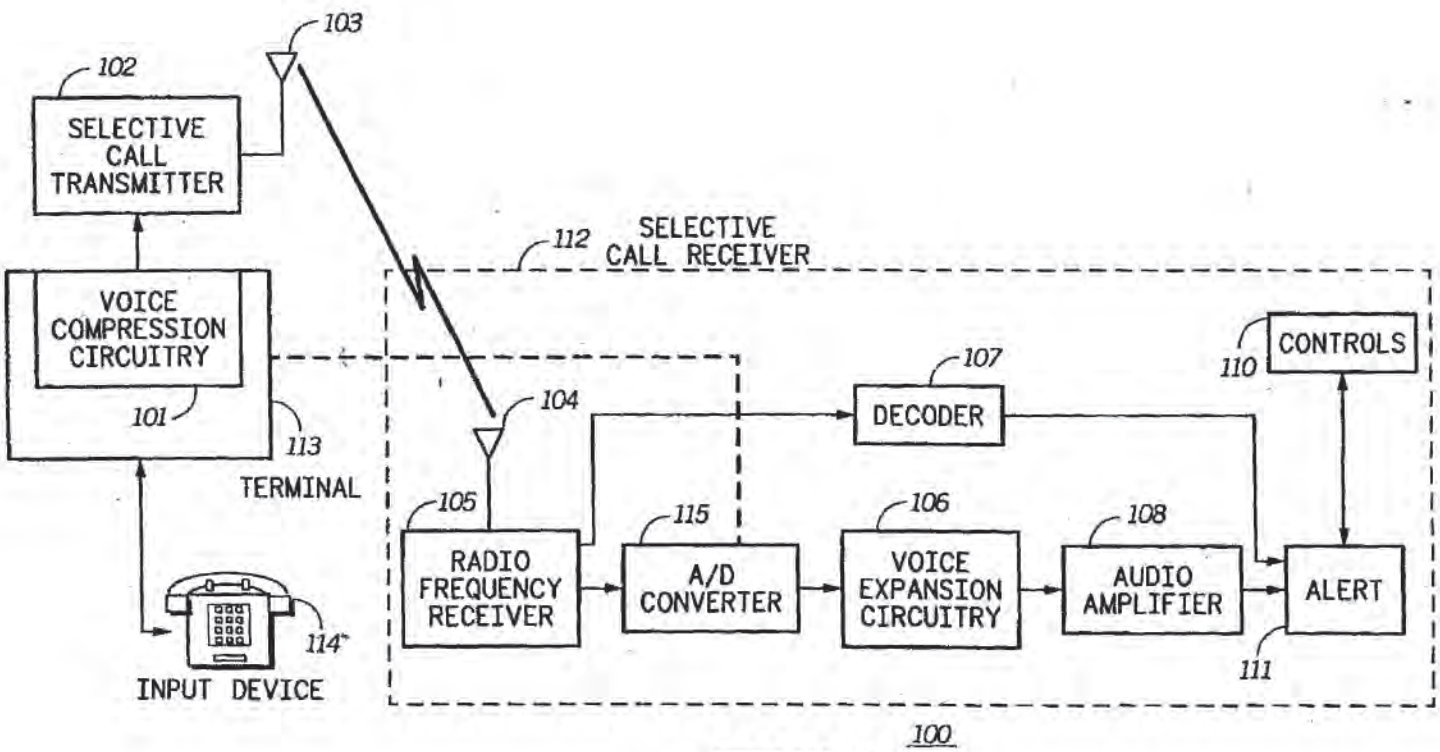

102
SELECTIVE CALL TRANSMITTER
103
VOICE COMPRESSION CIRCUITRY
101
113
TERMINAL
114
INPUT DEVICE
112
SELECTIVE CALL RECEIVER
104
105
RADIO FREQUENCY RECEIVER
115
A/D CONVERTER
107
DECODER
106
VOICE EXPANSION CIRCUITRY
108
AUDIO AMPLIFIER
110
CONTROLS
ALERT
111
100

FIG.1

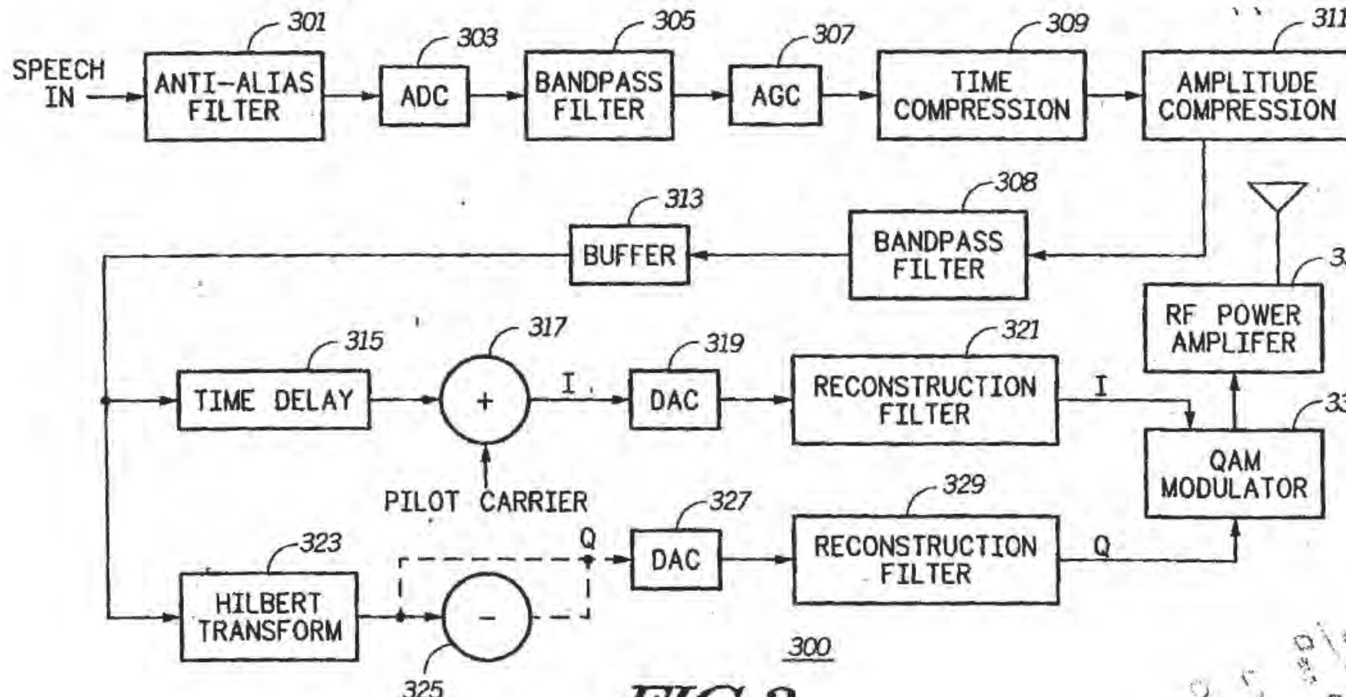

SPEECH IN
ANTI-ALIAS FILTER
301
ADC
303
BANDPASS FILTER
305
AGC
307
TIME COMPRESSION
309
AMPLITUDE COMPRESSION
311
BANDPASS FILTER
308
BUFFER
313
RF POWER AMPLIFER
335
TIME DELAY
315
+
317
I
DAC
319
RECONSTRUCTION FILTER
321
I
QAM MODULATOR
333
PILOT CARRIER
DAC
327
RECONSTRUCTION FILTER
329
Q
Q
HILBERT TRANSFORM
323
−
325
300

FIG.3

2/14

PRINT OF DRAWINGS AS ORIGINALLY FILED

APPROVED BY DRAFTSMAN | O.G. FIG. | CLASS

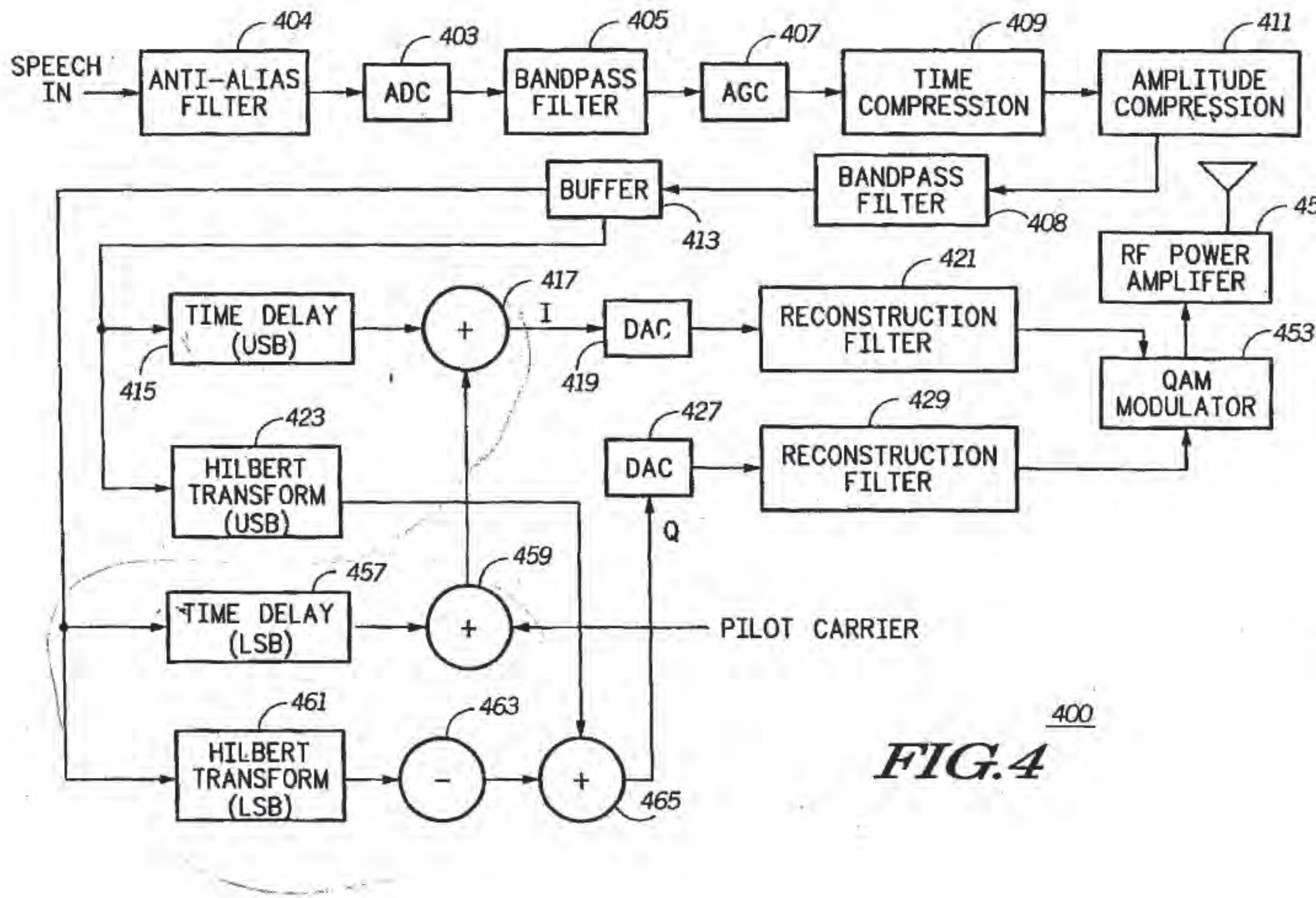


FIG.4

3/14

AS ORIGINALLY FILED

08/704450
~~08/395747~~

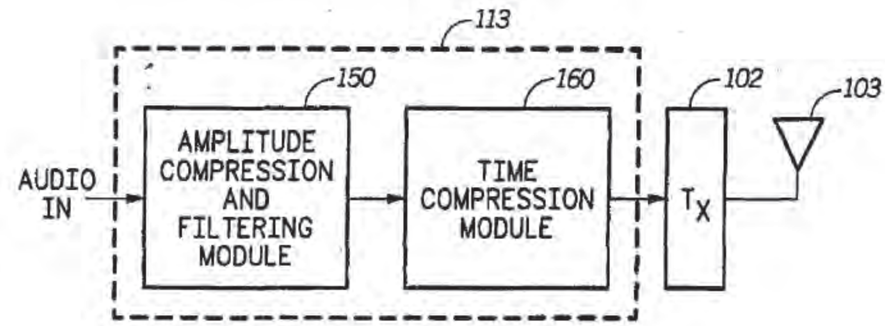


FIG.2

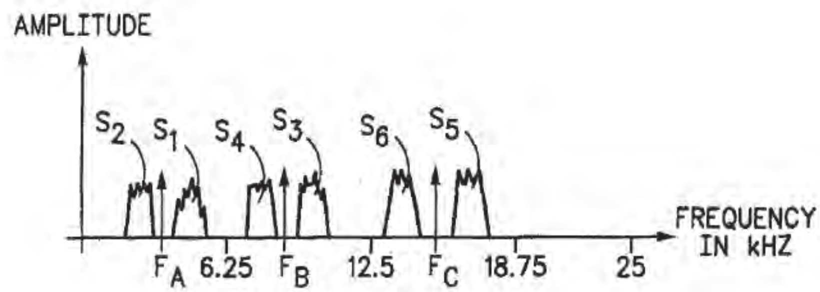


FIG.6

PRINT OF DRAWINGS
AS ORIGINALLY FILED

08/764656
~~08/395747~~

5/14

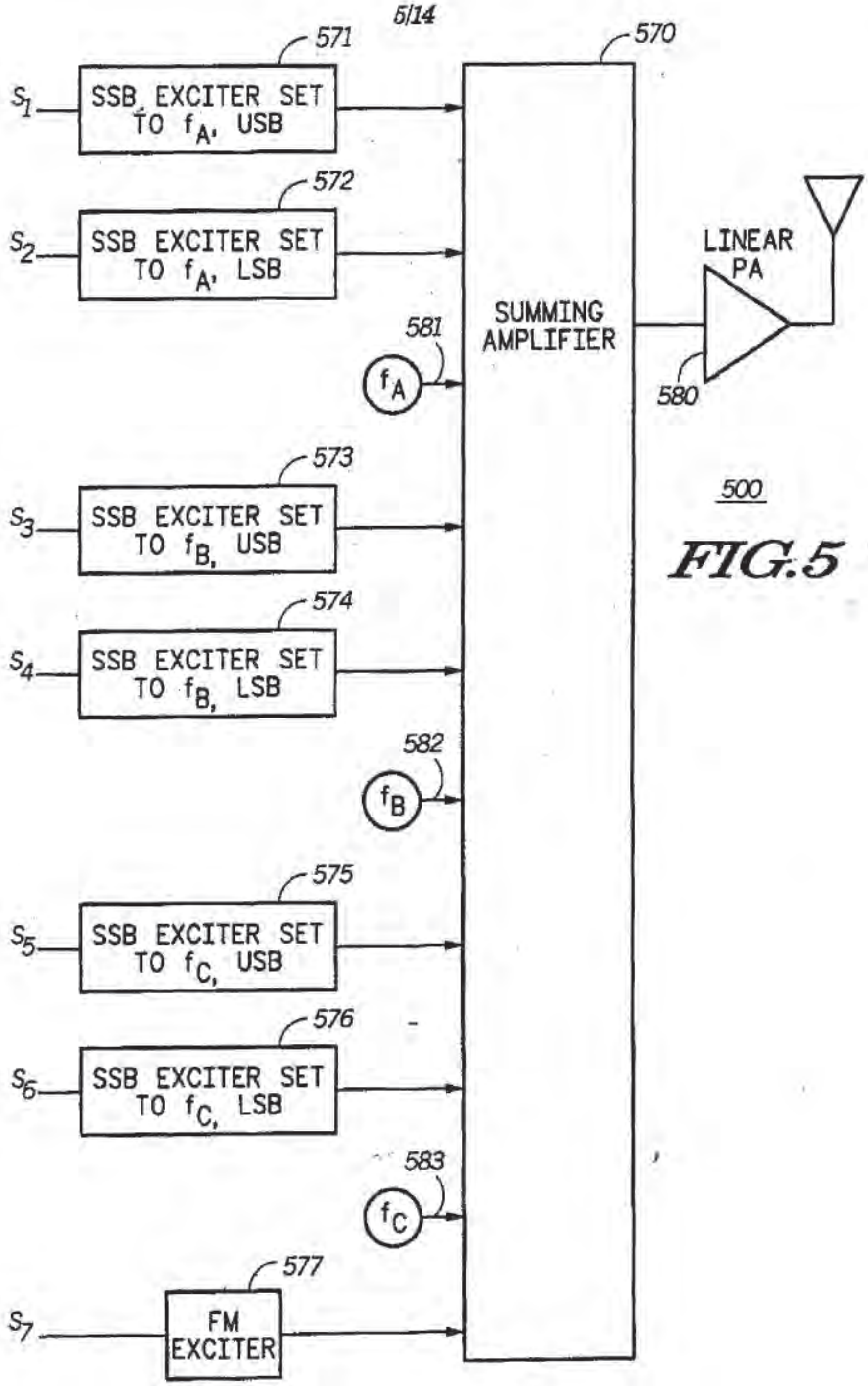


FIG.5

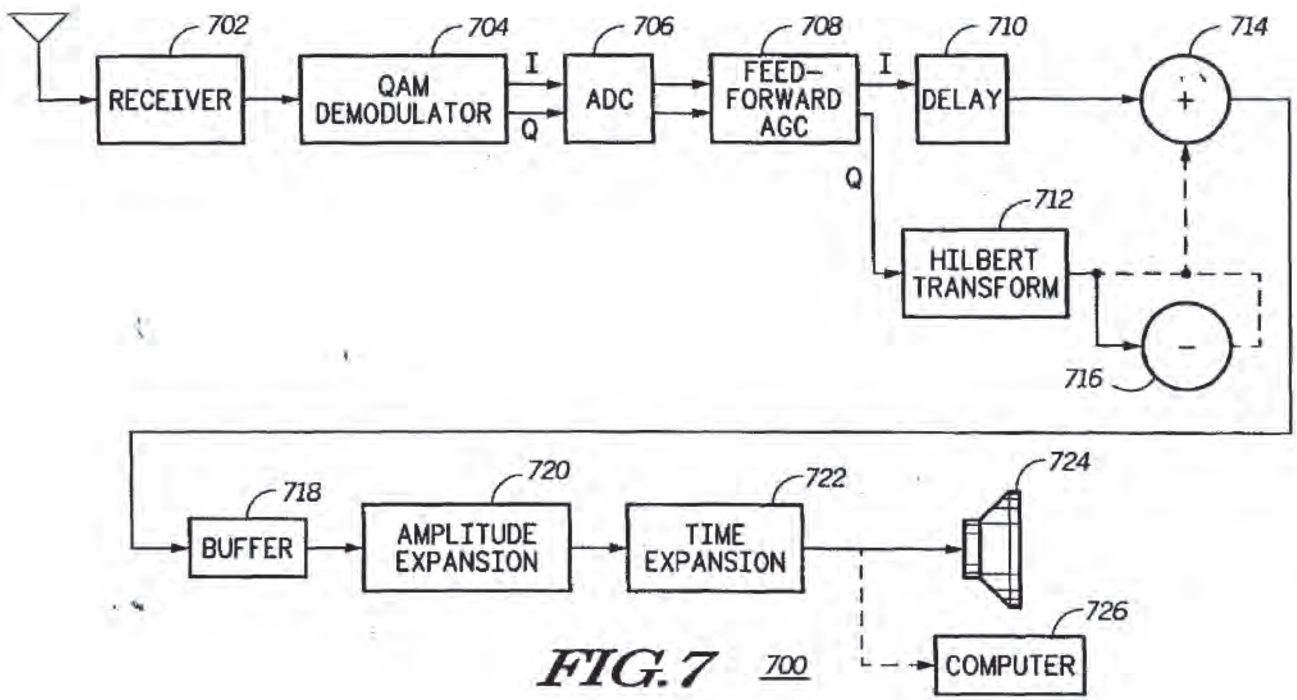

702
RECEIVER
704
QAM DEMODULATOR
I
Q
706
ADC
708
FEED-FORWARD AGC
I
Q
710
DELAY
714
+
712
HILBERT TRANSFORM
716
-
718
BUFFER
720
AMPLITUDE EXPANSION
722
TIME EXPANSION
724
726
COMPUTER

FIG. 7 700

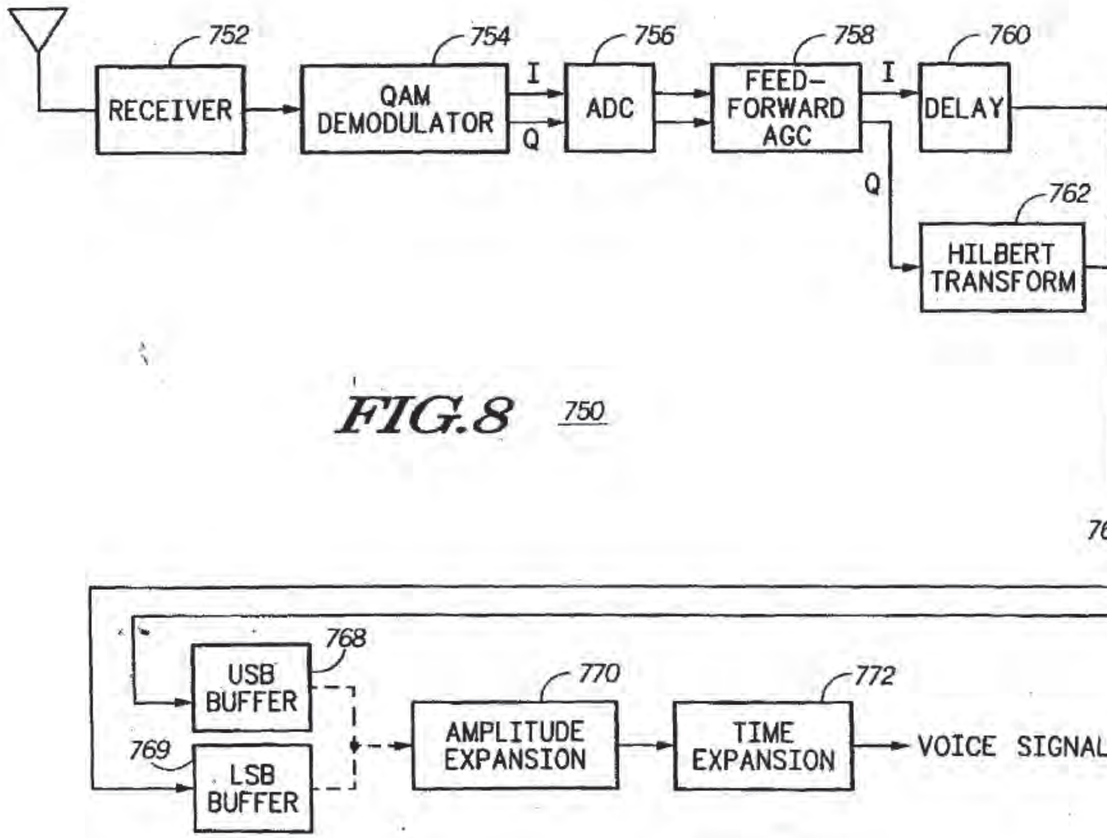

RECEIVER
752
QAM DEMODULATOR
754
I
Q
ADC
756
FEED-FORWARD AGC
758
I
Q
DELAY
760
HILBERT TRANSFORM
762
764
766
765
USB BUFFER
768
LSB BUFFER
769
AMPLITUDE EXPANSION
770
TIME EXPANSION
772
VOICE SIGNAL

FIG.8 750

7/14

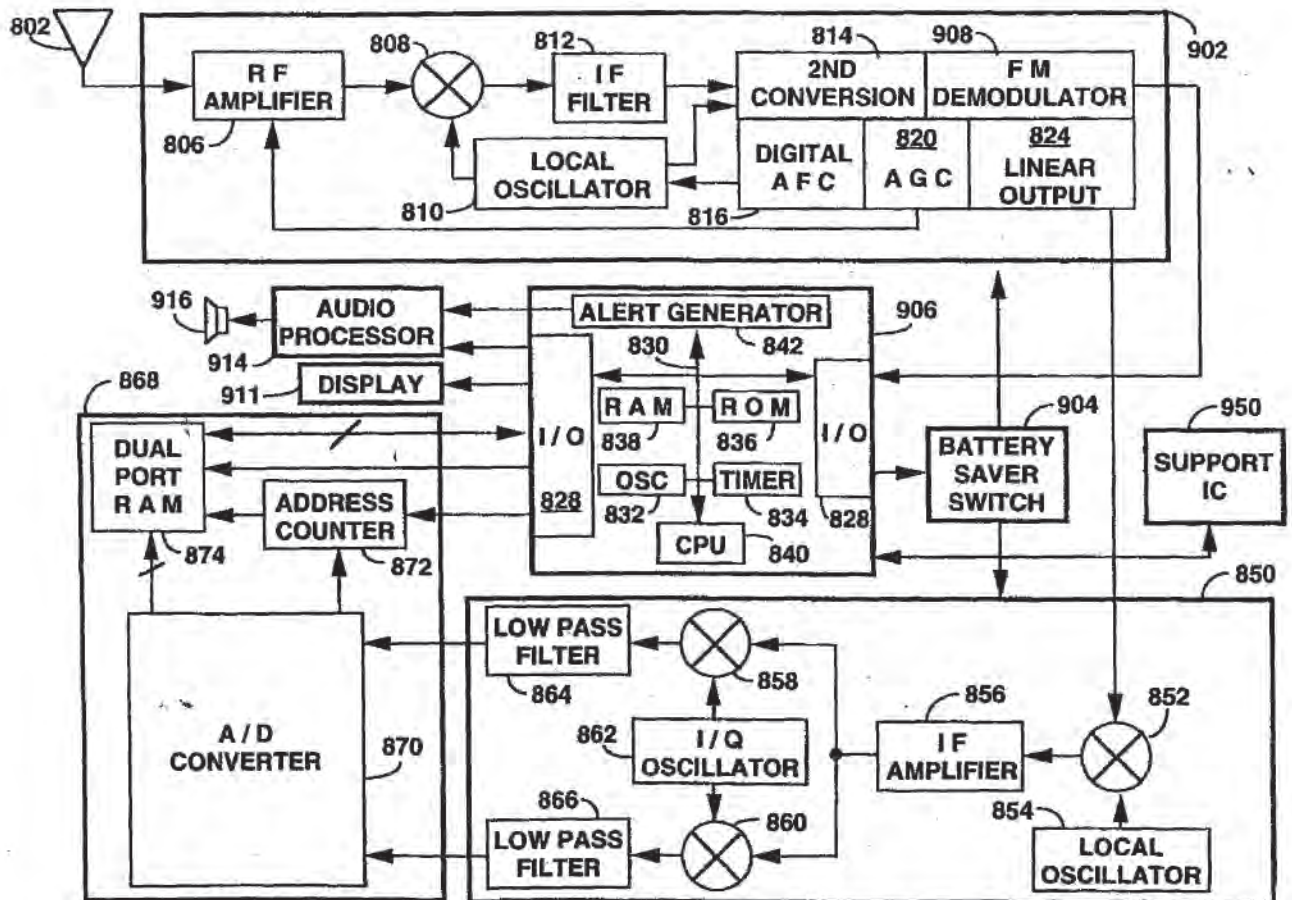

802
RF AMPLIFIER
806
808
810
812
IF FILTER
LOCAL OSCILLATOR
814
2ND CONVERSION
908
FM DEMODULATOR
902
DIGITAL AFC
816
820
AGC
824
LINEAR OUTPUT
916
AUDIO PROCESSOR
914
911
DISPLAY
ALERT GENERATOR
906
830
842
RAM
838
ROM
836
I/O
828
OSC
832
TIMER
834
CPU
840
868
DUAL PORT RAM
874
ADDRESS COUNTER
872
904
BATTERY SAVER SWITCH
950
SUPPORT IC
850
LOW PASS FILTER
864
858
856
IF AMPLIFIER
852
862
I/Q OSCILLATOR
A/D CONVERTER
870
866
860
854
LOCAL OSCILLATOR

FIG. 9 900

08/764656

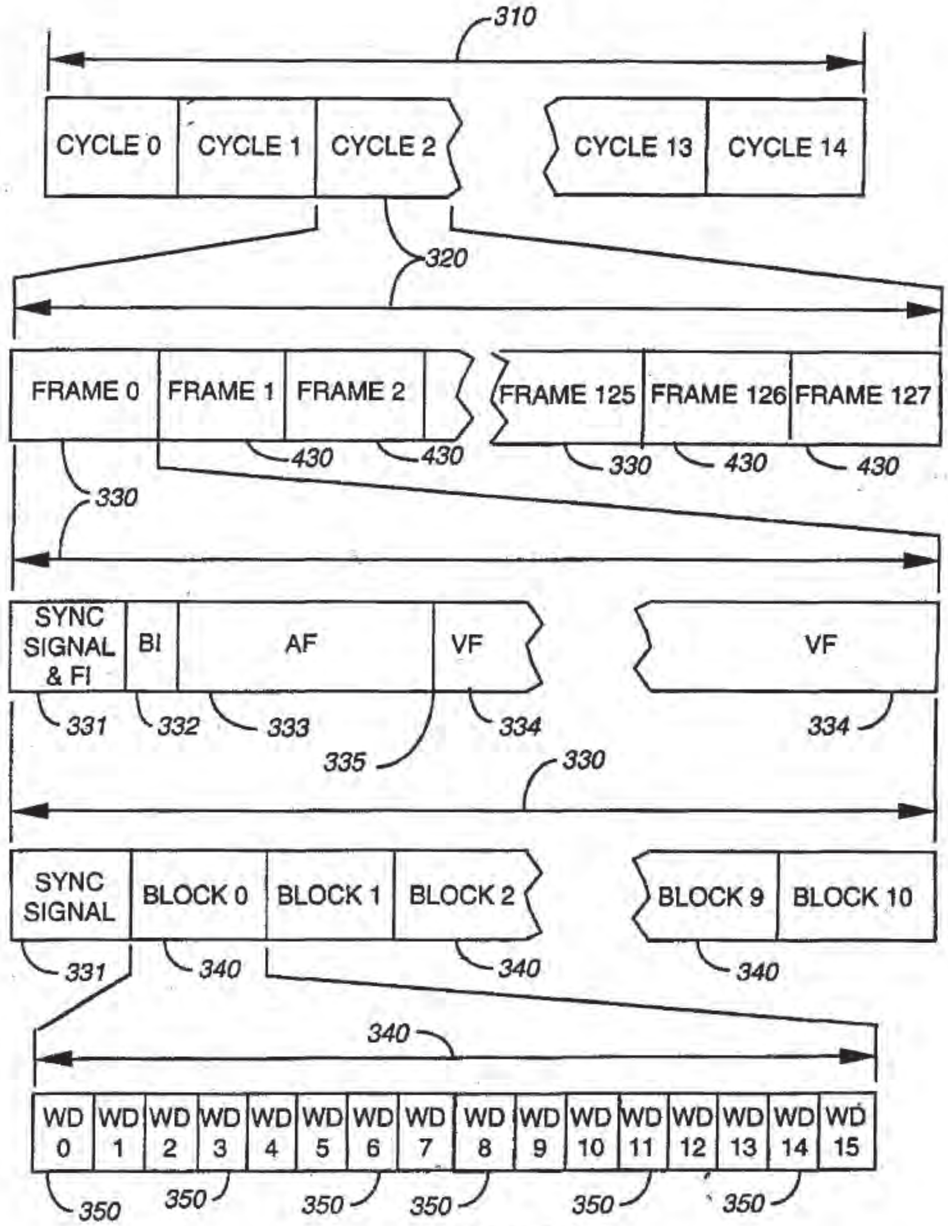


FIG.10

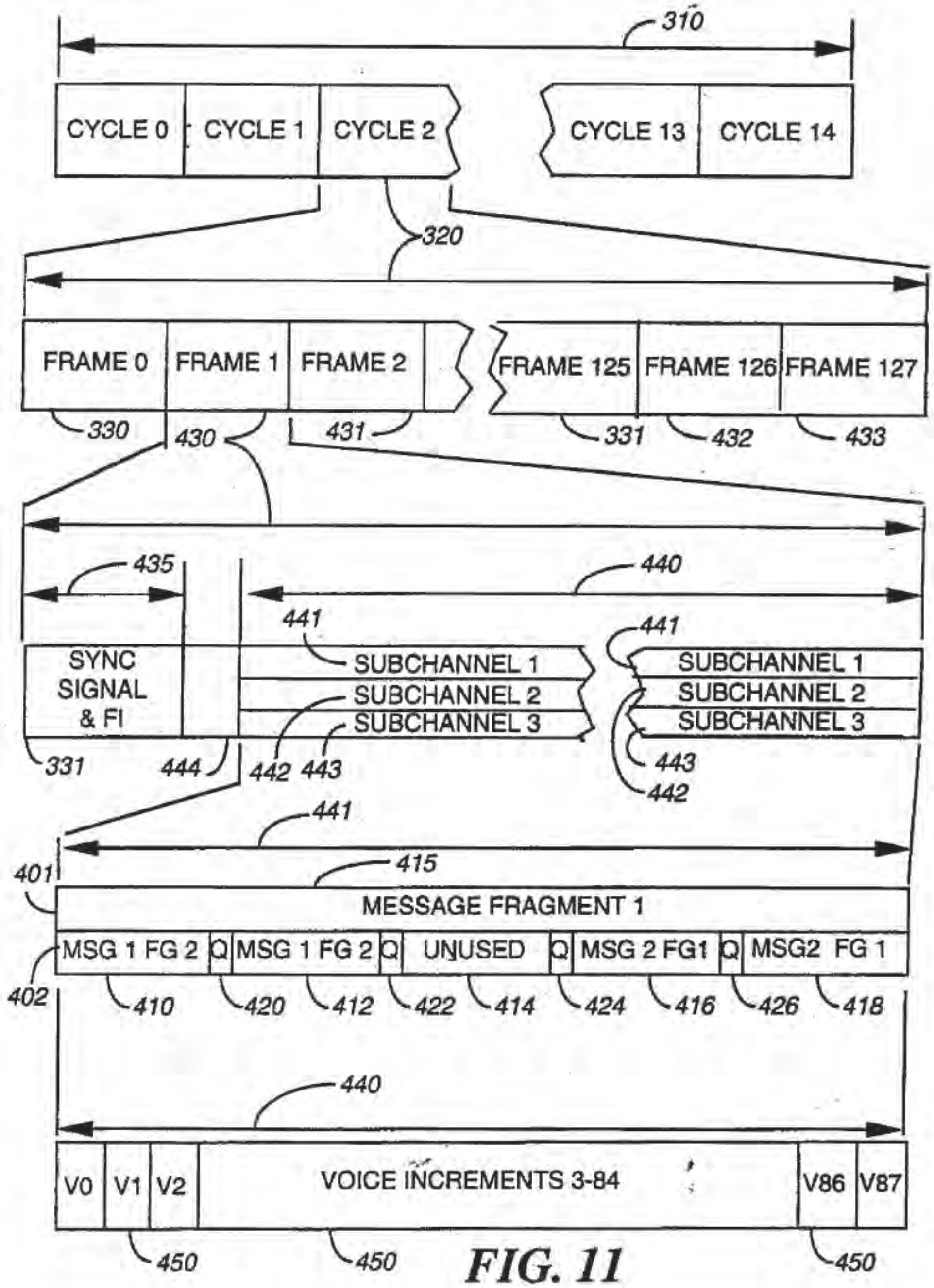

310
CYCLE 0
CYCLE 1
CYCLE 2
CYCLE 13
CYCLE 14
320
FRAME 0
FRAME 1
FRAME 2
FRAME 125
FRAME 126
FRAME 127
330
430
431
331
432
433
435
440
441
SYNC SIGNAL & FI
SUBCHANNEL 1
SUBCHANNEL 2
SUBCHANNEL 3
331
444
442
443
441
415
401
MESSAGE FRAGMENT 1
MSG 1 FG 2
Q
MSG 1 FG 2
Q
UNUSED
Q
MSG 2 FG1
Q
MSG2 FG 1
402
410
420
412
422
414
424
416
426
418
440
V0
V1
V2
VOICE INCREMENTS 3-84
V86
V87
450

FIG. 11

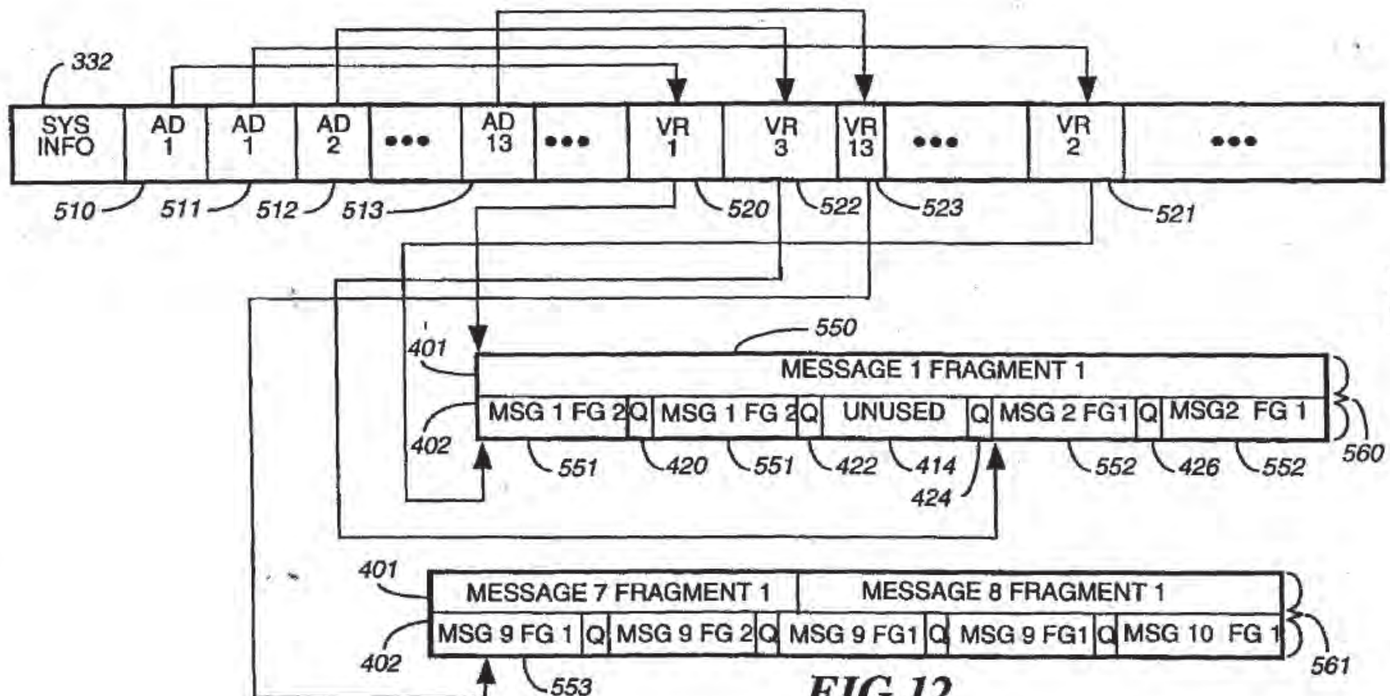

332
SYS INFO
AD 1
AD 1
AD 2
AD 13
VR 1
VR 3
VR 13
VR 2
510
511
512
513
520
522
523
521
550
401
MESSAGE 1 FRAGMENT 1
MSG 1 FG 2
Q
MSG 1 FG 2
Q
UNUSED
Q
MSG 2 FG1
Q
MSG2 FG 1
402
551
420
551
422
414
424
552
426
552
560
401
MESSAGE 7 FRAGMENT 1
MESSAGE 8 FRAGMENT 1
MSG 9 FG 1
Q
MSG 9 FG 2
Q
MSG 9 FG1
Q
MSG 9 FG1
Q
MSG 10 FG 1
402
553
561

FIG.12

PRINT OF DRAWINGS
AS ORIGINALLY FILED

08/395747

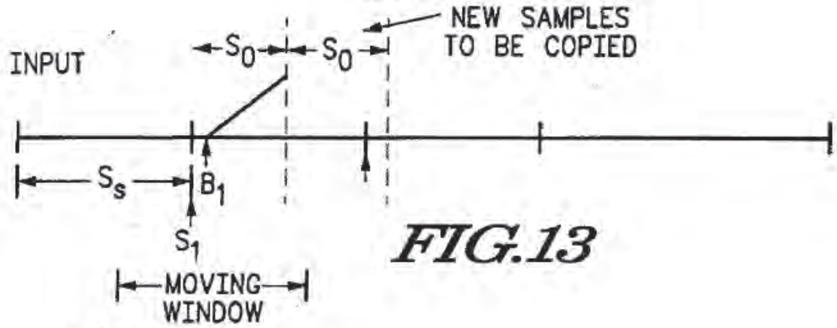


FIG.13

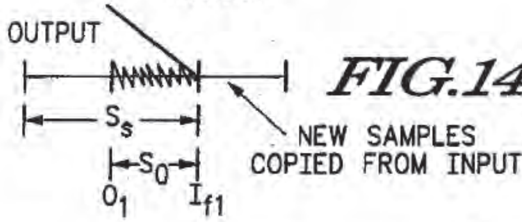


FIG.14

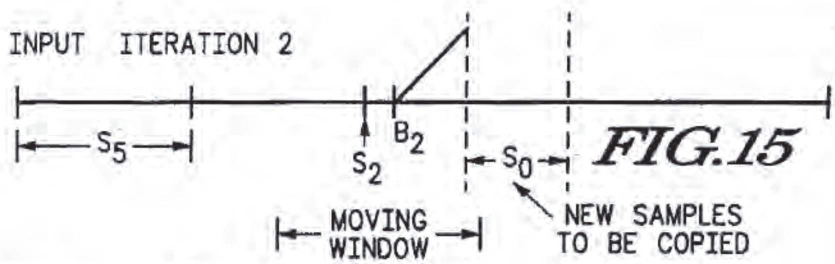


FIG.15

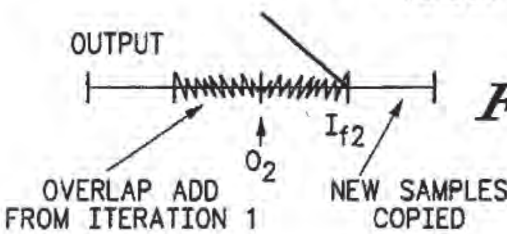


FIG.16

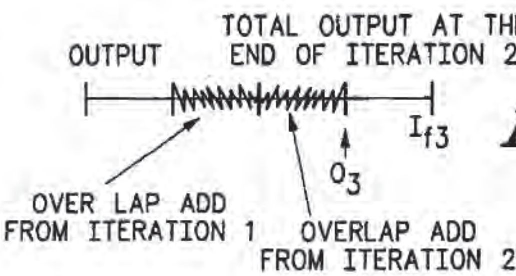


FIG.17

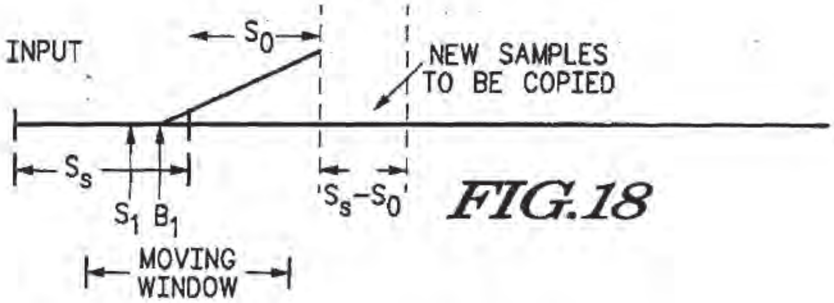


FIG.18

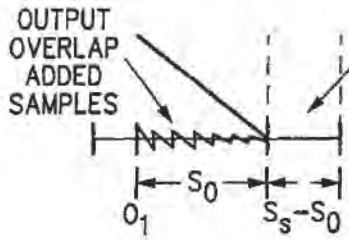


FIG.19

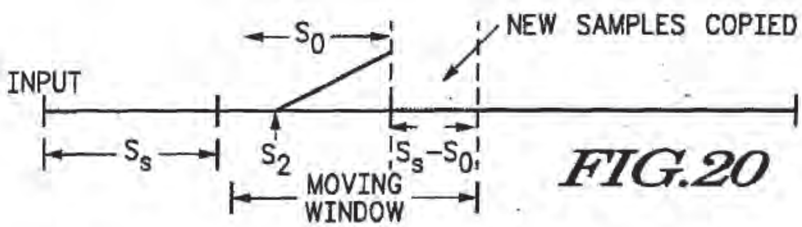


FIG.20

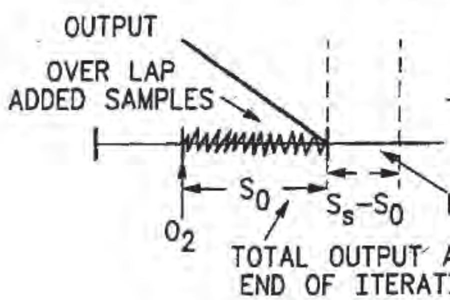


FIG.21

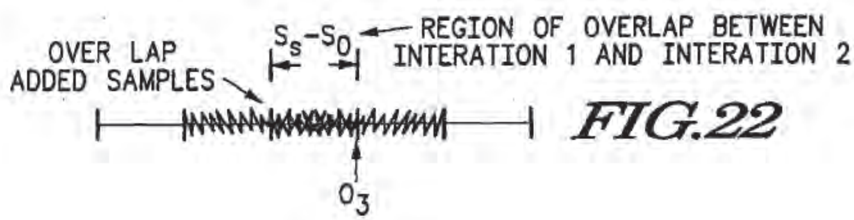


FIG.22

08/395747

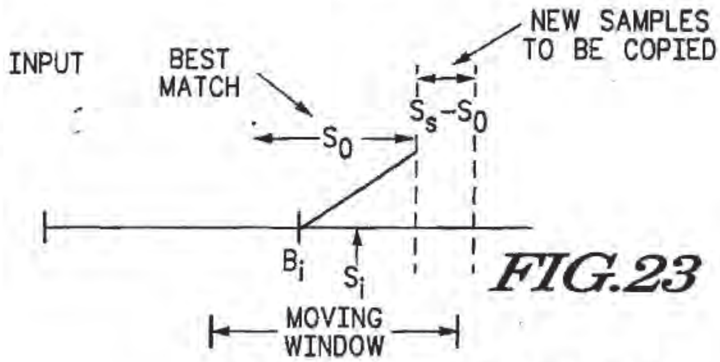


FIG. 23

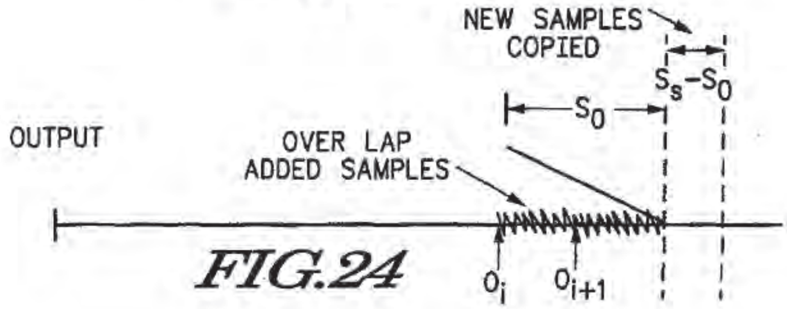


FIG. 24

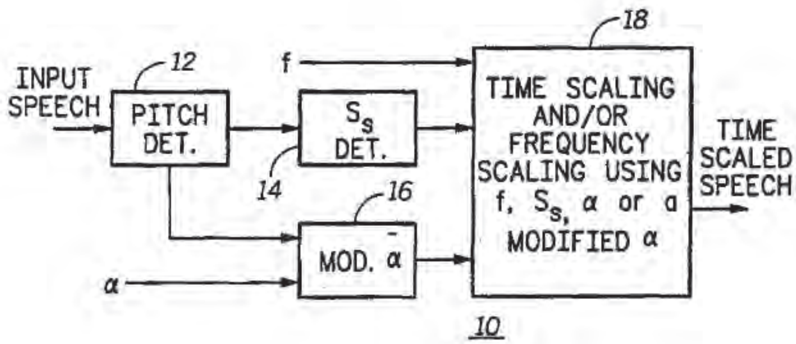


FIG. 25

MOTOROLA, INC.
PT0600U
LEITCH, et al.

APPROVED	O.G. FIG. 1	
BY	CLASS	SUBCLASS
DRAFTSMAN	364	514R

5689440

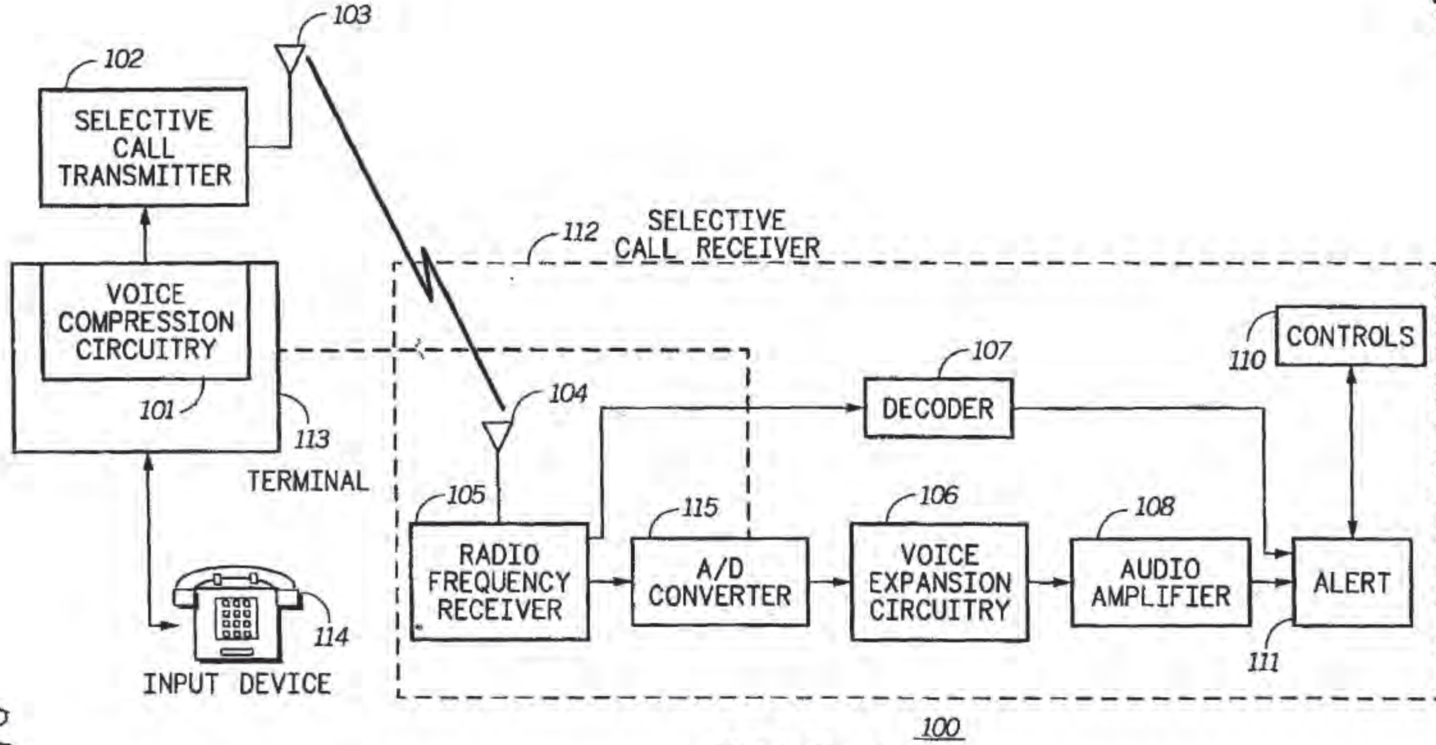


FIG.1

25 figs.

08/764 056

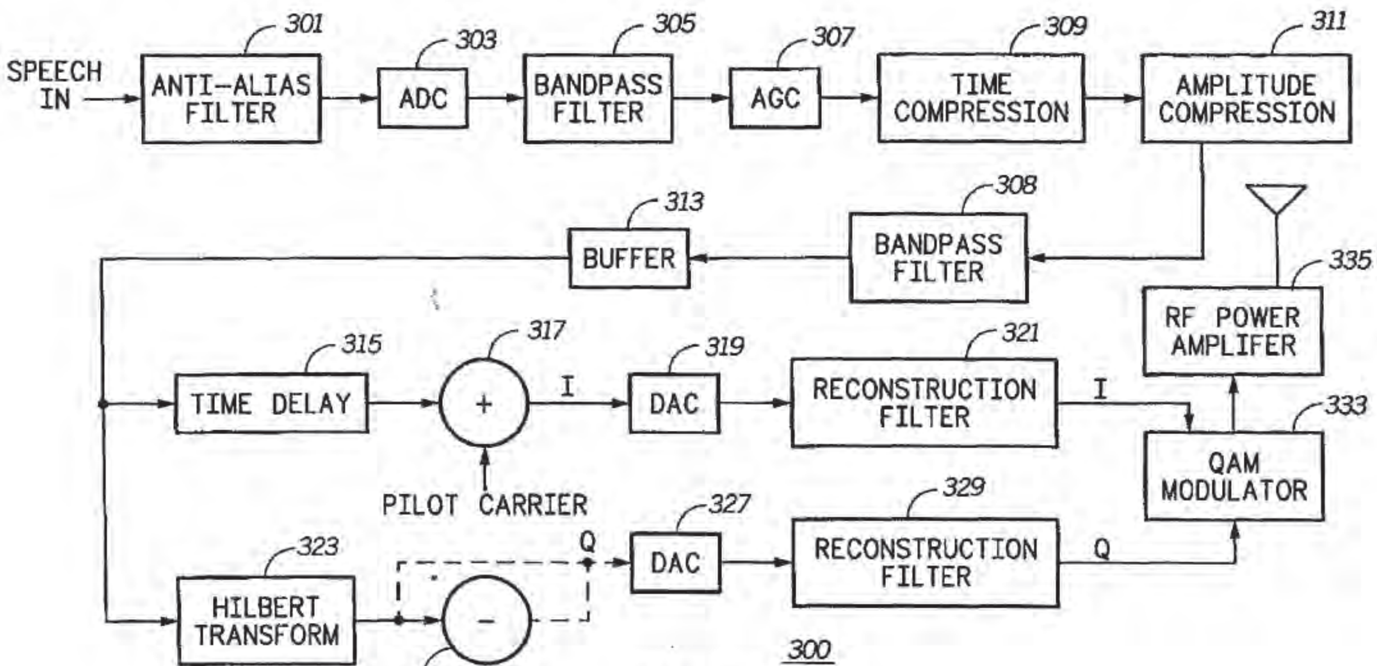


FIG.3

2/14

08/764,686

APPROVED BY DRAFTSMAN	O.G. FIG.	
	CLASS	SUBCLASS

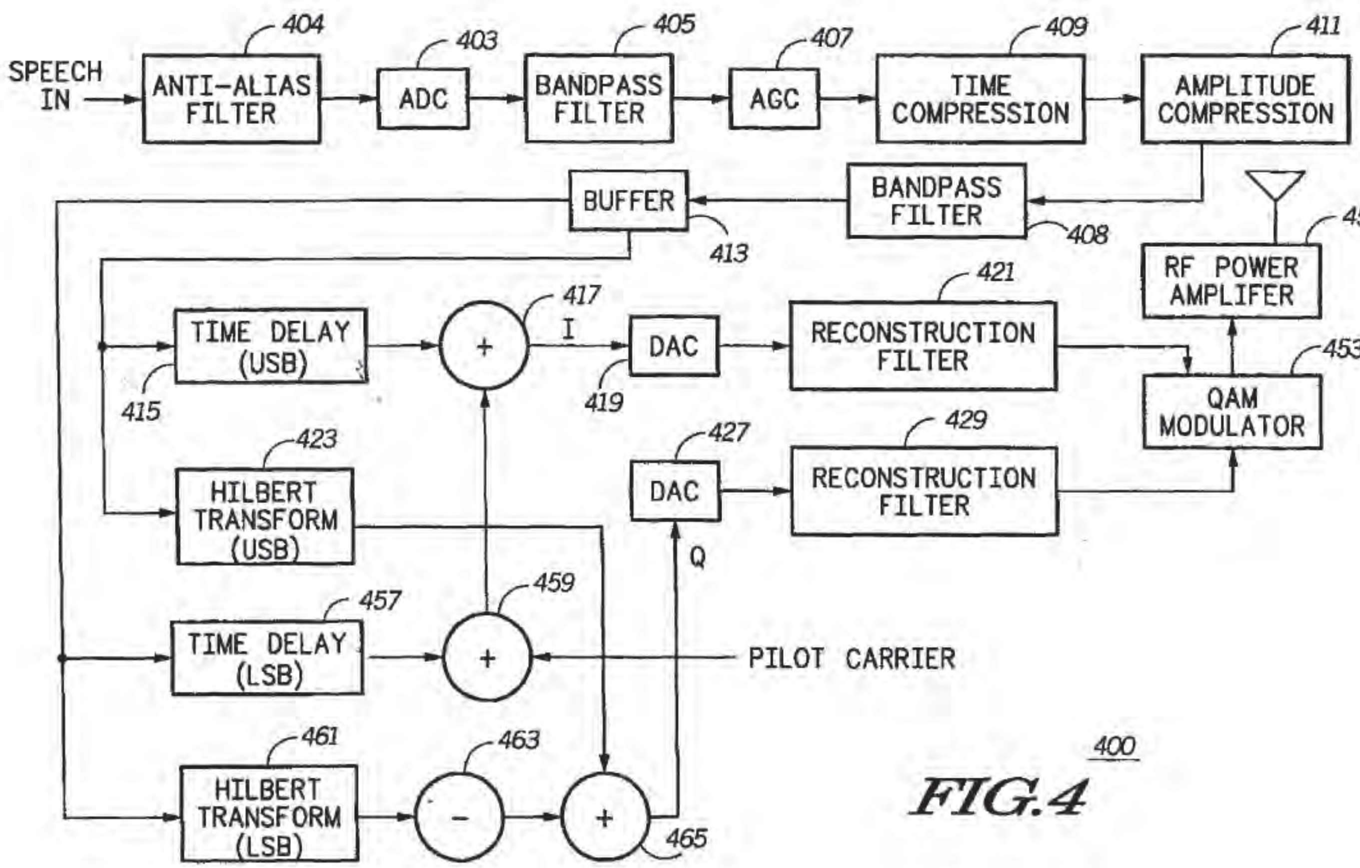


FIG.4

3/14

08/704656

08/764656
~~08/395747~~

O.G. FIG.		
APPROVED	BY	DRAFTSMAN
CLASS	SUBCLASS	

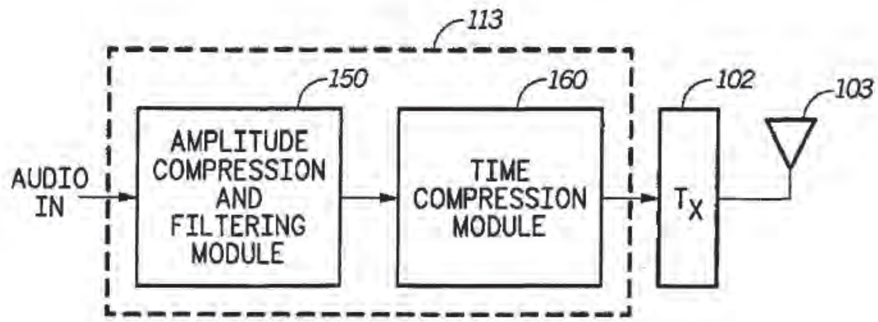


FIG.2

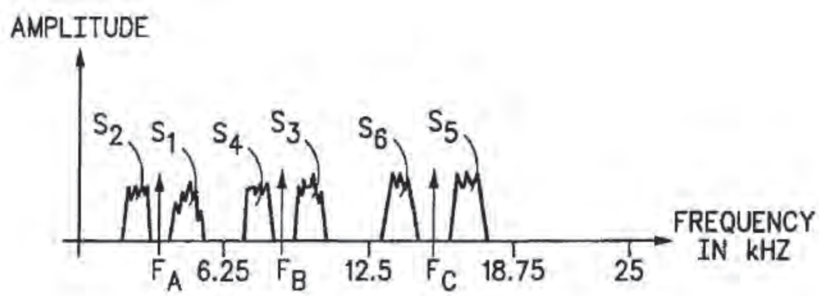
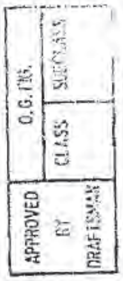
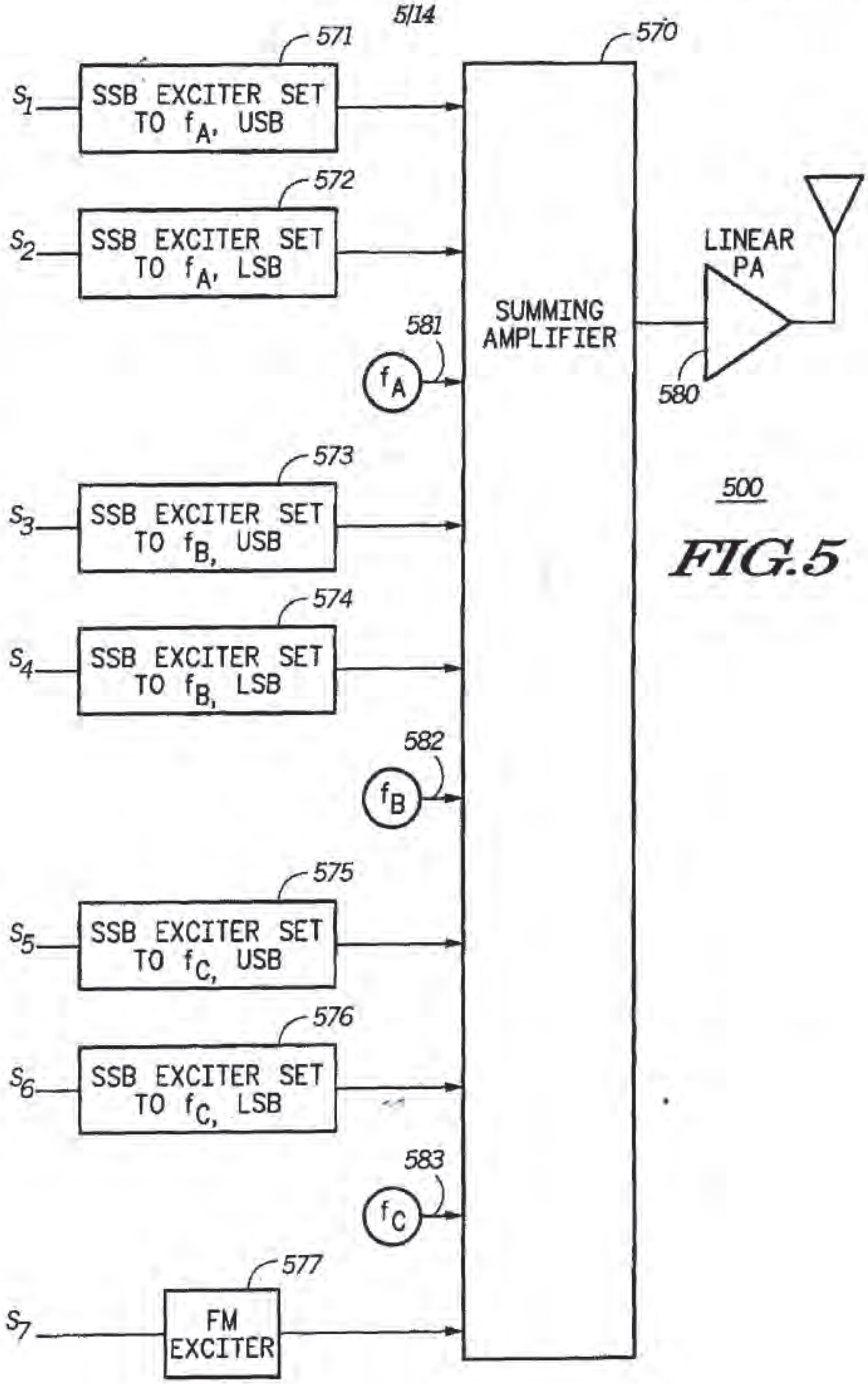


FIG.6

~~08/395747~~
08/764656

5/14





500

FIG.5

APPROVED BY DRAFTSMAN	O.G. FIG.	
	CLASS	SUBCLASS

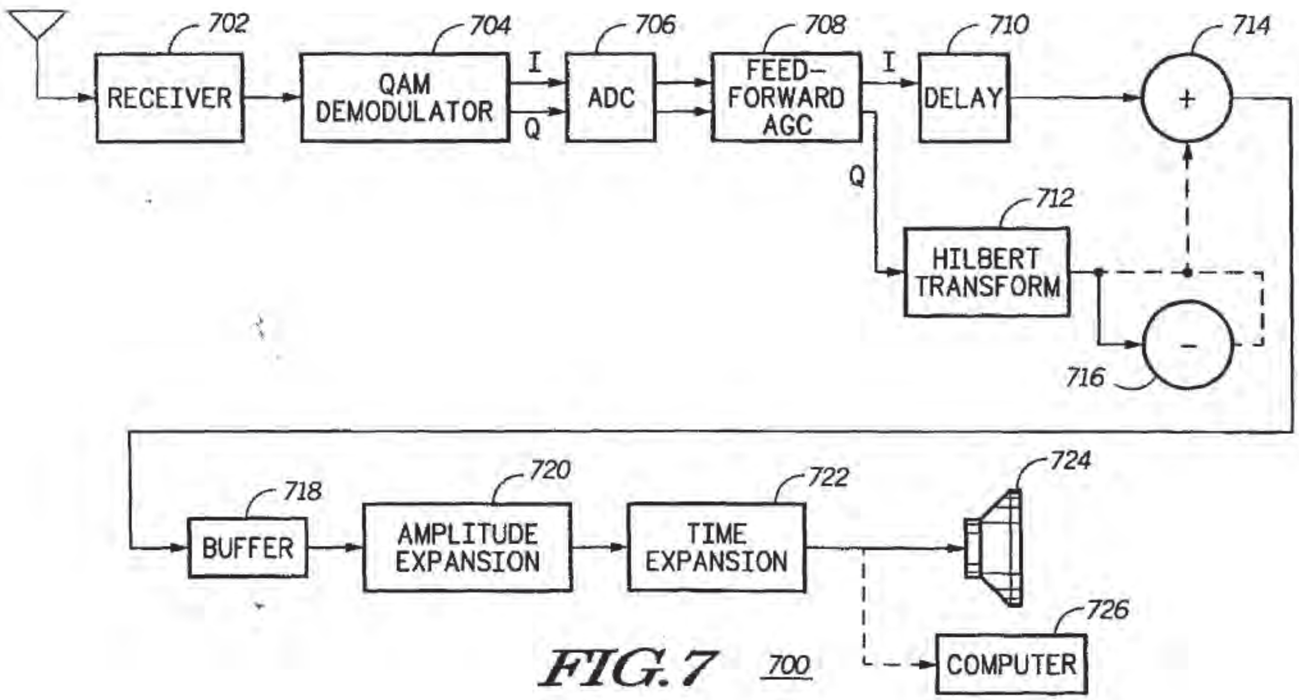


FIG. 7 700

6/14

~~08/395747~~
08/764656

APPROVED BY DRAFTSMAN	O.G. FIG.	
	CLASS	SUBCLASS

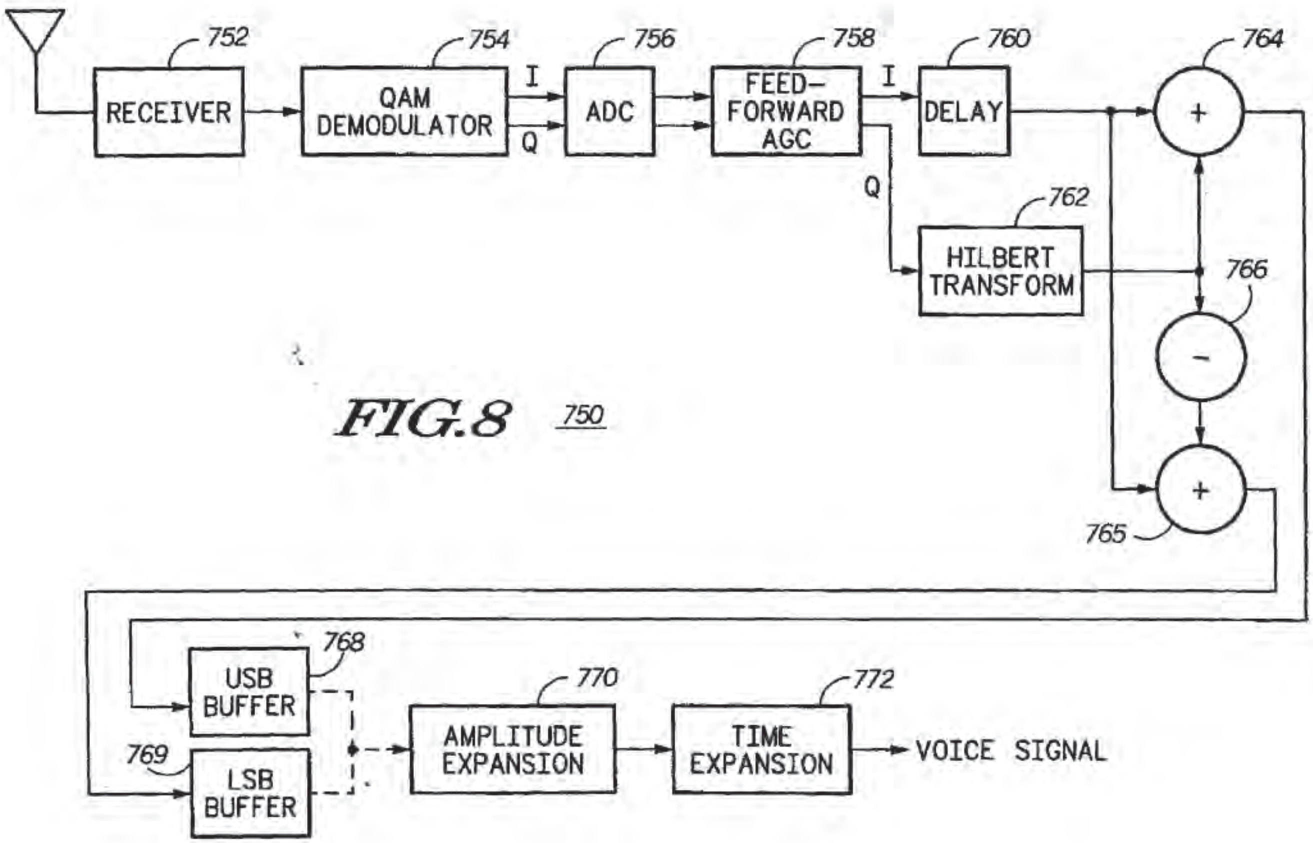


FIG.8 750

7/14

08/764,056

APPROVED BY DRAFTSMAN	O.G. FIG.	
	CLASS	SUBCLASS

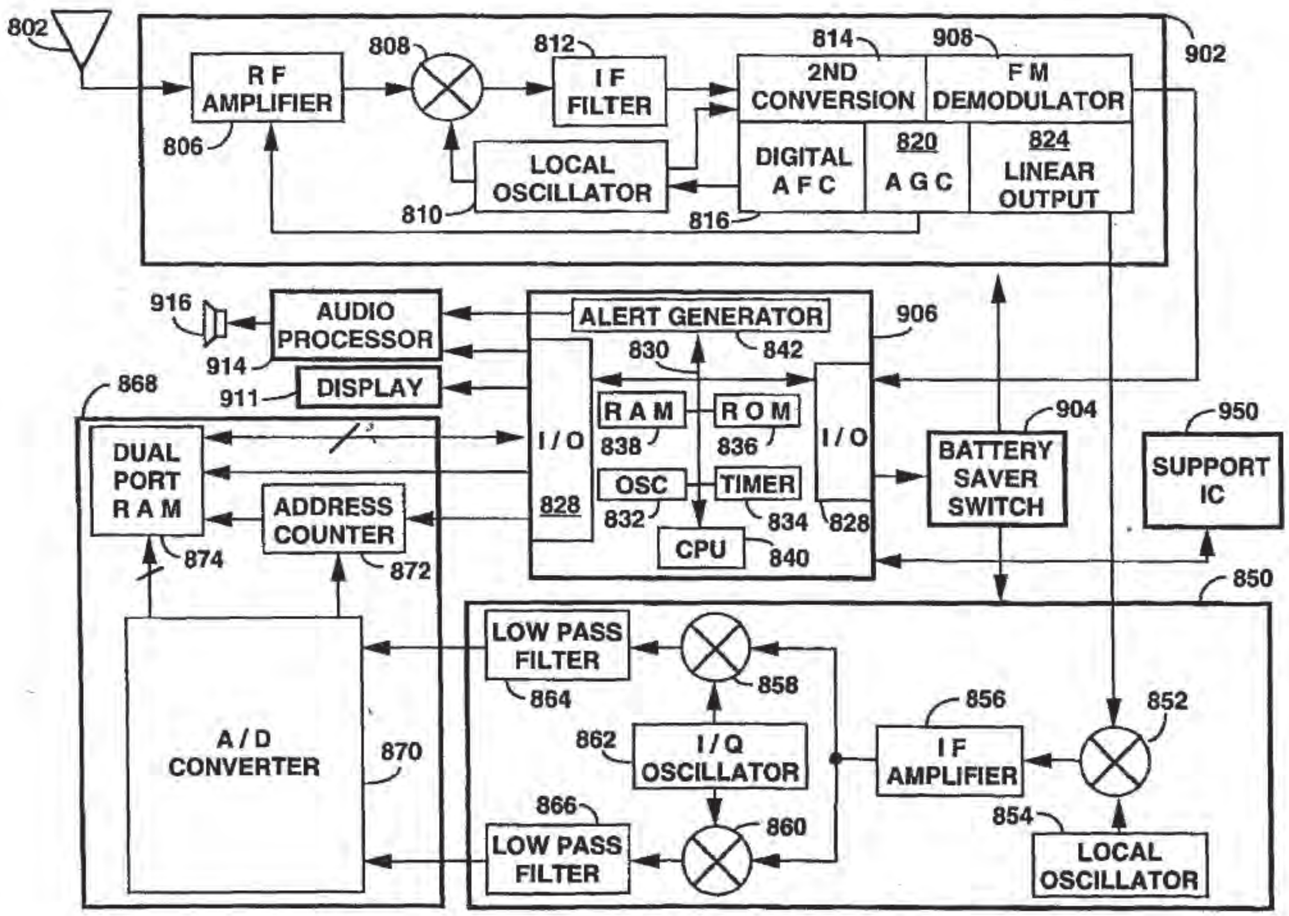


FIG. 9 900

8/14

08/764656

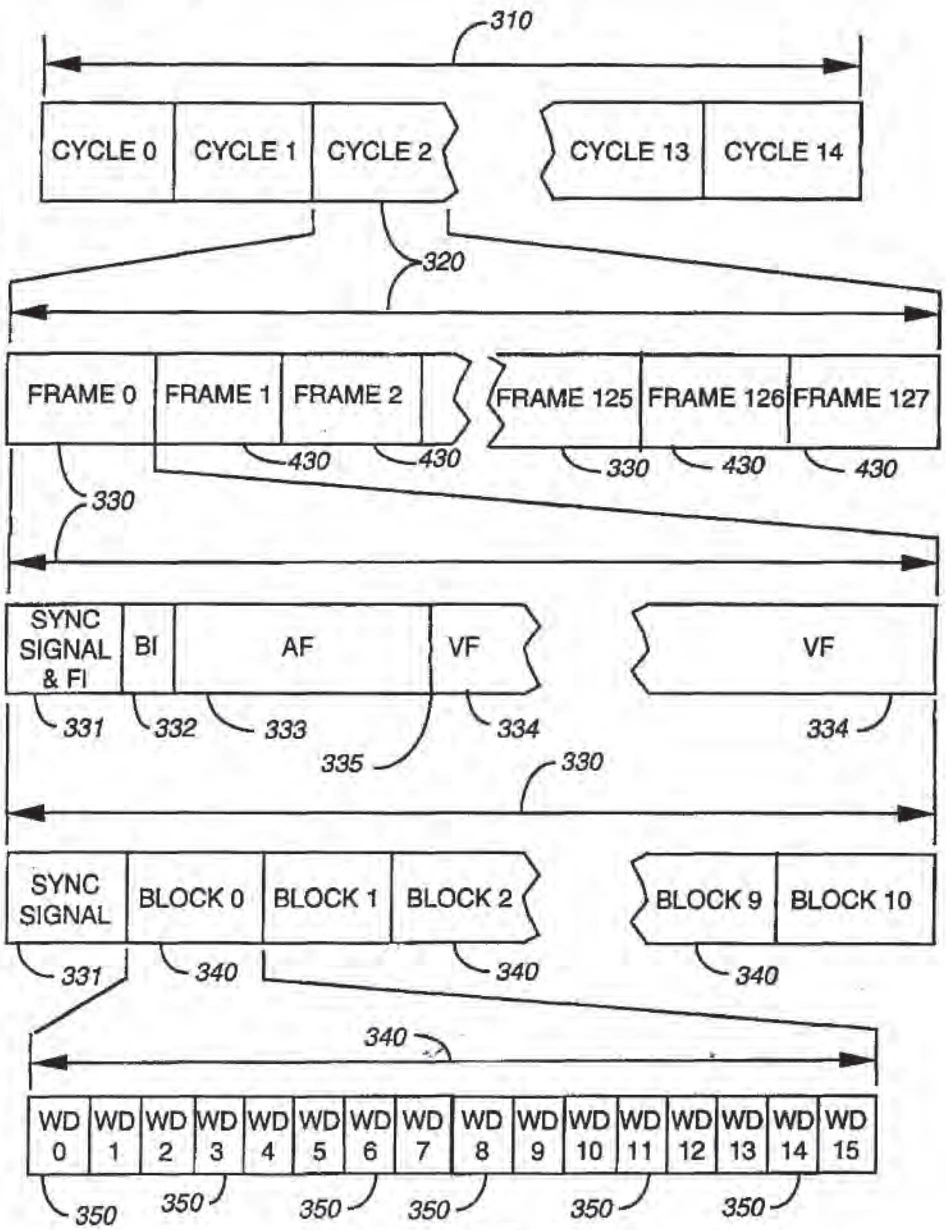

310
CYCLE 0
CYCLE 1
CYCLE 2
CYCLE 13
CYCLE 14
320
FRAME 0
FRAME 1
FRAME 2
FRAME 125
FRAME 126
FRAME 127
430
430
330
430
430
330
SYNC SIGNAL & FI
BI
AF
VF
VF
331
332
333
334
334
335
330
SYNC SIGNAL
BLOCK 0
BLOCK 1
BLOCK 2
BLOCK 9
BLOCK 10
331
340
340
340
340
WD 0
WD 1
WD 2
WD 3
WD 4
WD 5
WD 6
WD 7
WD 8
WD 9
WD 10
WD 11
WD 12
WD 13
WD 14
WD 15
350
350
350
350
350
350

FIG.10

~~08/395747~~ 08/764656

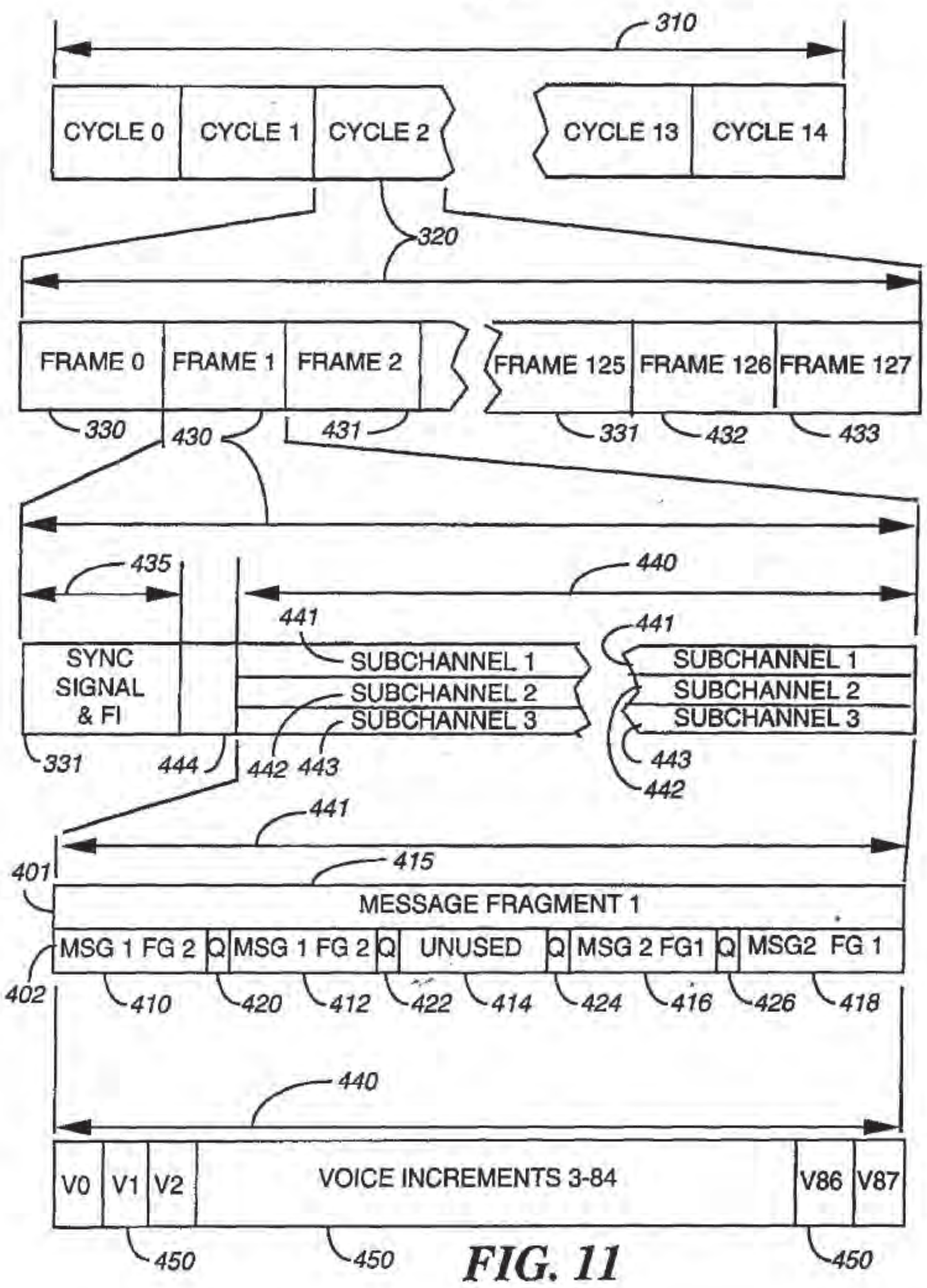


FIG. 11

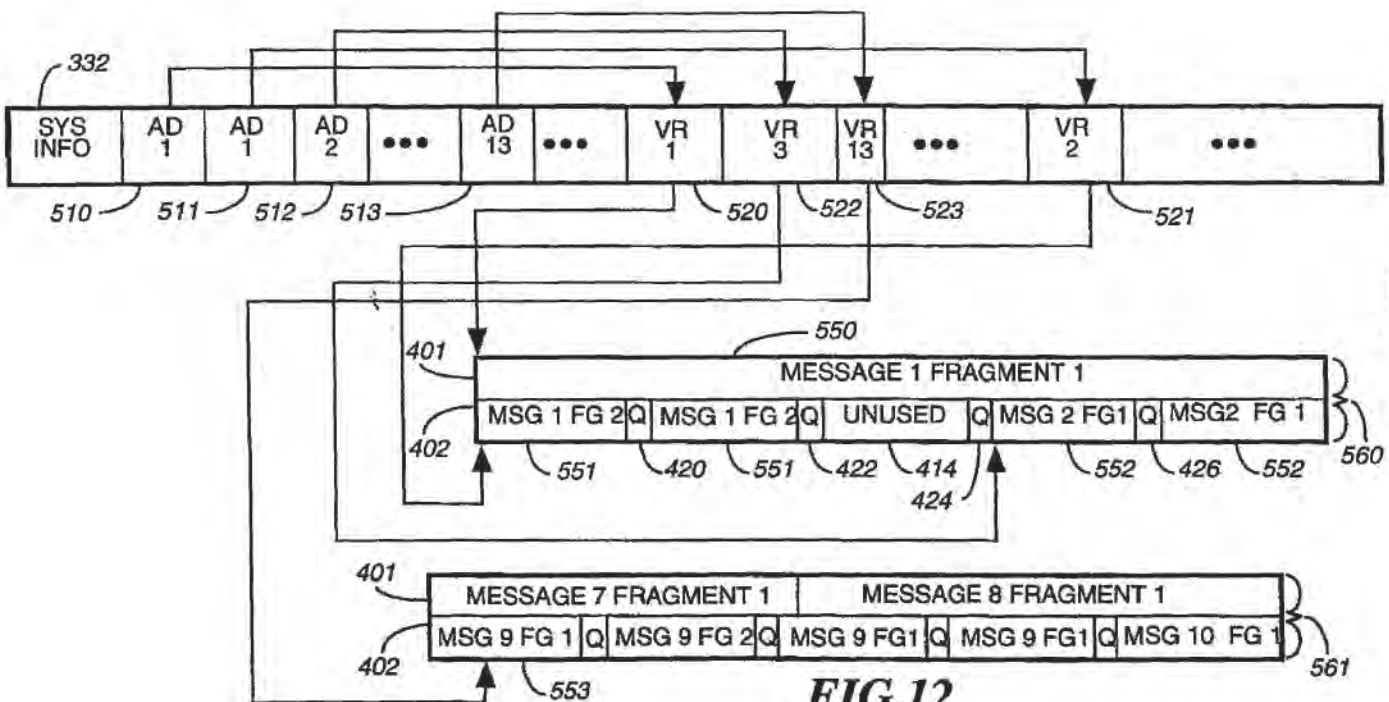

332
SYS INFO
AD 1
AD 1
AD 2
AD 13
VR 1
VR 3
VR 13
VR 2
510
511
512
513
520
522
523
521
550
401
MESSAGE 1 FRAGMENT 1
402
MSG 1 FG 2
Q
MSG 1 FG 2
Q
UNUSED
Q
MSG 2 FG1
Q
MSG2 FG 1
560
551
420
551
422
414
424
552
426
552
401
MESSAGE 7 FRAGMENT 1
MESSAGE 8 FRAGMENT 1
402
MSG 9 FG 1
Q
MSG 9 FG 2
Q
MSG 9 FG1
Q
MSG 9 FG1
Q
MSG 10 FG 1
561
553

FIG.12

08/764656
08/395747

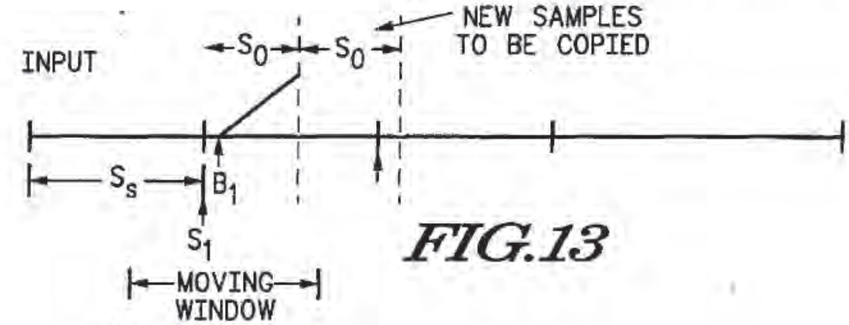


FIG.13

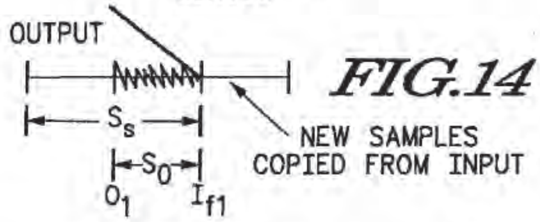


FIG.14

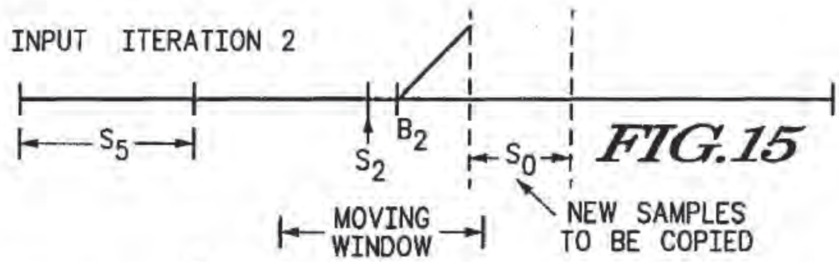


FIG.15

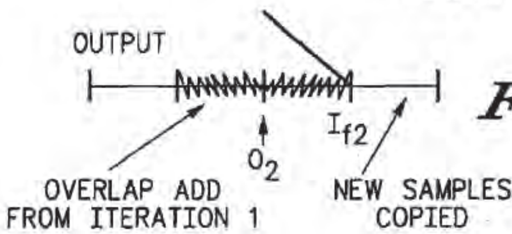


FIG.16

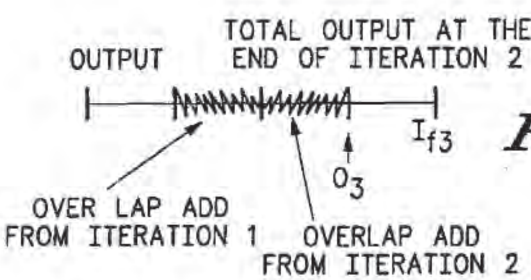


FIG.17

08/395747
09/764650

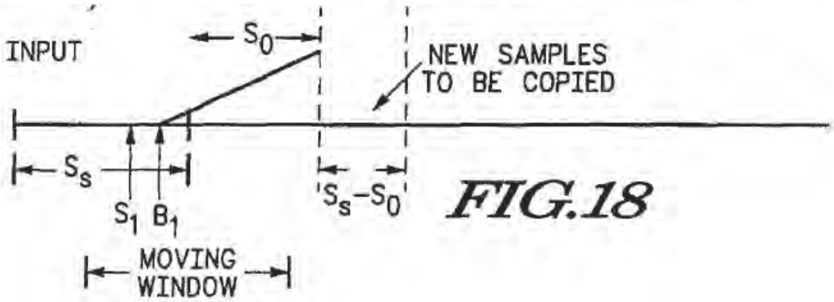


FIG.18

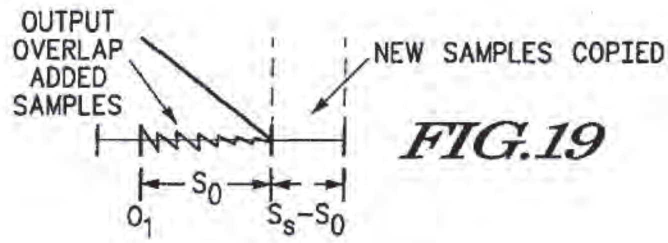


FIG.19

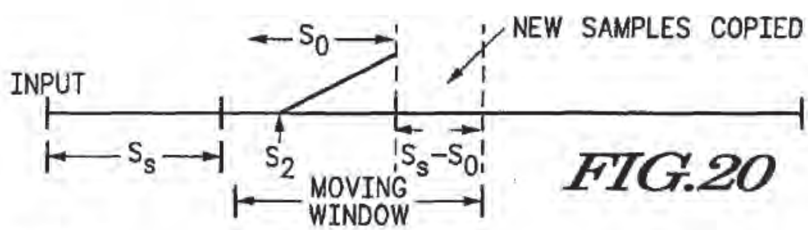


FIG.20

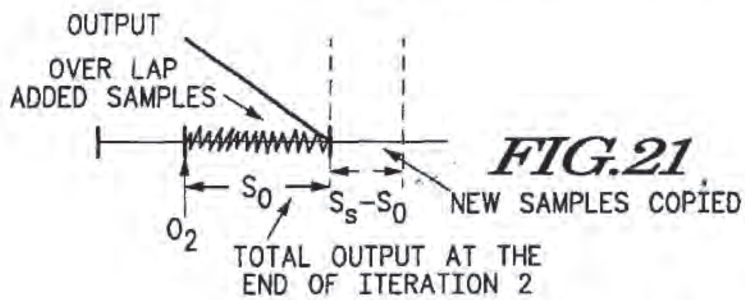


FIG.21

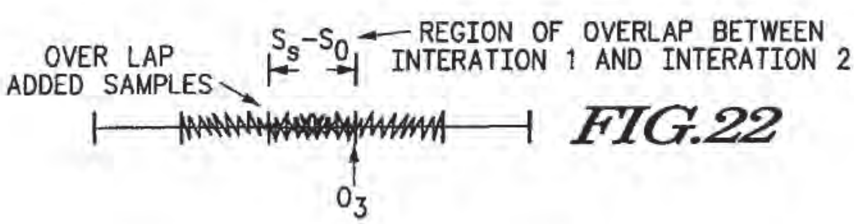


FIG.22

08/395747

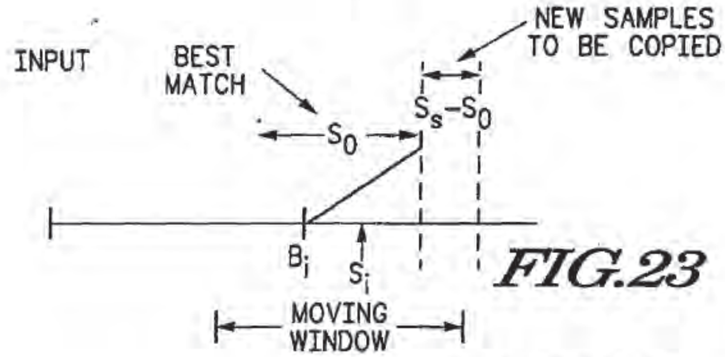


FIG.23

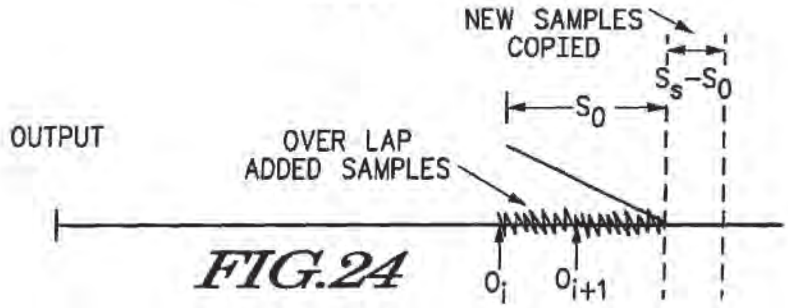


FIG.24

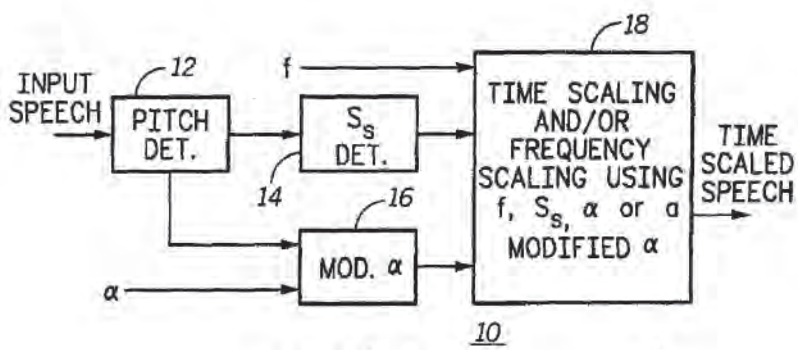


FIG.25

14

~~08/395747~~ 08/704656

Docket No.: PT00600U

Inventors: Clifford D. Leitch
Robert J. Schwendeman
Kazimierz Siwiak
William J. Kuznicki
Sunil Satyamurti

VOICE COMPRESSION METHOD AND APPARATUS IN A COMMUNICATION SYSTEM

Ins. Q1

Technical Field

This invention relates generally to voice compression techniques, and more particularly a method and apparatus of voice compression using efficient bandwidth utilization and time compression techniques.

Background

Voice message paging is not economically feasible for large paging systems with current technology. The air time required for a voice page is much more than that required for a tone, numeric or alphanumeric page. With current technology, voice paging service would be economically prohibitive in comparison to tone, numeric or alphanumeric paging with less than ideal voice quality reproduction. Another constraint in limiting voice message paging is the bandwidth and the present methods of utilizing the bandwidth of paging channels. In comparison, the growth of alphanumeric paging has been constrained by the limited access to a keyboard input device for sending alphanumeric messages to a paging terminal, either in the form of a personal keyboard or a call to an operator center. A voice system overcomes these entry issues since a caller can simply pick up a telephone, dial access numbers, and speak a message. Further, none of the present voice paging systems take advantage of Motorola's new high speed paging protocol structure, also known as FLEX™.

Existing voice paging systems lack many of the FLEX™ protocol advantages including high battery saving ratios, multiple channel scanning capability, mixing of modes such as voice with data, acknowledge-back paging (allowing for return receipts to the calling party), location finding capability, system and frequency reuse, particularly in large metropolitan areas, and range extension through selective re-transmission of missed message portions.

With respect to the aspect of paging involving time-scaling of voice signals and to other applications such as dictation and voice mail, current methods of time-scaling lack the ideal combinations of providing adequate speech quality and flexibility that allows a designer to optimize the application within the constraints given. Thus, there exists a need for a voice communication system that is economically feasible and flexible in allowing optimization within a given configuration, and more particularly with respect to paging applications, that further retains many of the advantages of Motorola's FLEX™ protocol.

Summary of the Invention

In one aspect, the present invention comprises a method for compressing a plurality of voice signals within a voice communication resource having a given bandwidth within a voice communication system. The method comprises the steps of subchanneling the voice communication resource while placing at least one of each of the plurality of voice signals on a subchannel and compressing the time of each of the voice signals within each of the subchannels, wherein these steps provide a compressed voice signal.

In another aspect of the present invention, a communication system using voice compression has at least one transmitter base station and a plurality of selective call receivers. The transmitter base station comprises an input device for receiving an audio signal, a processing device for compressing the audio signal using a time-scale compression technique and a single side band modulation technique to provide a processed signal and a quadrature amplitude modulator for the subsequent transmission of the processed signal. Each of the plurality of selective call receivers comprises a selective call receiver module for receiving the transmitted processed signal, a processing device for demodulating the received processed signal using a single side band demodulation technique and a time-scale expansion technique to provide a reconstructed signal, and an amplifier for amplifying the reconstructed signal into an reconstructed audio signal.

In another aspect of the present invention, a selective call receiver for receiving compressed voice signals, comprises a selective call receiver module for receiving a transmitted processed signal, a processing device for demodulating the received processed signal using a single side band demodulation technique and a time-scale expansion technique to provide a reconstructed signal, and an amplifier for amplifying the reconstructed signal into an reconstructed audio signal.

In yet another aspect of the present invention, a paging base station for transmitting selective call signals on a communication resource having a predetermined bandwidth, comprises, an input device for receiving a plurality of audio signals, a device for subchannelizing the communication resource into a predetermined number of subchannels, an amplitude compression and filtering module for each subchannel for compressing the amplitude of the respective audio signal and filtering the respective audio signal, a time compression module for compression of the time of the respective audio signal for each subchannel, and a quadrature amplitude modulator for the subsequent transmission of the processed signal.

Brief Description of the Drawings

FIG. 1 is a block diagram of a voice communication system in accordance with the present invention.

FIG. 2 is a block diagram of a base station transmitter in accordance with the present invention.

FIG. 3 is an expanded electrical block diagram of the base station transmitter in accordance with the present invention.

FIG. 4 is an expanded electrical block diagram of another base station transmitter in accordance with the present invention.

FIG. 5 is block diagram of a speech processing, encoding, and modulation portion of a base station transmitter in accordance with the present invention.

FIG.6 is a spectrum analyzer output of a 6 single-sideband signal transmitter in accordance with the present invention.

4

FIG. 7 is an expanded electrical block diagram of a selective call receiver in accordance with the present invention.

FIG. 8 is an expanded electrical block diagram of another selective call receiver in accordance with present invention.

FIG. 9 is an expanded electrical block diagram of another selective call receiver in accordance with present invention.

FIG. 10 is a timing diagram showing the transmission format of an outbound signaling protocol in accordance with the present invention.

FIG. 11 is another timing diagram showing the transmission format of an outbound signaling protocol including details of a voice frame in accordance with the present invention.

FIG. 12 is another timing diagram illustrating a control frame and two analog frames of the outbound signaling protocol in accordance with the present invention.

FIGs. 13-17 illustrate timing diagrams for several iterations of the WSOLA time-scaling (compression) method in accordance with the present invention.

FIGs. 18-22 illustrate timing diagrams for several iterations of the WSOLA-SD time-scaling (compression) method in accordance with the present invention.

FIGs. 23-24 illustrate timing diagrams for iterations of the WSOLA-SD time-scaling (expansion) method in accordance with the present invention.

FIG. 25 illustrates a block diagram of the overall WSOLA-SD time scaling method in accordance with the present invention.

Detailed Description of the Preferred Embodiment

Referring to FIG. 1, a communication system illustrative of the voice compression and expansion techniques of the present invention are shown in a block diagram of the selective call system 100 which comprises an input device for receiving an audio signal such as telephone 114 (or other input device such as a computer) from which voice based selective calls are initiated for transmission to selective call receivers in the system 100. Each selective call entered through the telephone 114 typically comprises (a) a receiver address of at least one of the selective call receivers in the system and (b) a voice message. The initiated selective calls are typically provided to a transmitter base station or a selective call terminal 113 for formatting and queuing. Voice compression circuitry 101 of the terminal 113 serves to compress the time length of the provided voice message (the detailed operation of such voice compression circuitry 101 is discussed in the following description of FIGs. 2, 3 and 4). Preferably, the voice compression circuitry 101 includes a processing device for compressing the audio signal using a time-scaling technique and a single sideband modulation technique to provide a processed signal. The selective call is then input to the selective call transmitter 102 where it is applied as modulation to a radio frequency signal which is sent over the air through an antenna 103. Preferably, the transmitter is a quadrature amplitude modulation transmitter for transmitting the processed signal.

An antenna 104 within a selective call receiver 112 receives the modulated, transmitted radio frequency signal and inputs it to a selective call receiver module or radio frequency receiver module 105 for receiving the processed signal or radio frequency signal, where the radio frequency signal is demodulated and the receiver address and the compressed voice message modulation are recovered. The compressed voice message is then provided to an analog to digital converter (A/D) 115. Preferably, the selective call receiver 112 includes a processing device for demodulating the received processed signal using a single sideband demodulation technique and a time-scaling expansion technique to provide a reconstructed signal. The compressed voice message is then provided to a voice expansion circuit 106 where the time length of the voice message is preferably expanded to the desired value (the detailed operation of such voice expansion circuitry 106 used in the present invention is discussed in the following description of FIGs. 7 and 8). The voice message is then provided to an amplifier such as audio amplifier 108 for the purpose of amplifying it to a reconstructed audio signal.

The demodulated receiver address is supplied from the radio frequency receiver 105 to a decoder 107. If the receiver address matches any of the receiver addresses stored in the decoder 107, an alert 111 is optionally activated, providing a brief sensory indication to the user of the selective call receiver 112 that a selective call has been received. The brief sensory indication may comprise an audible signal, a tactile signal such as a vibration, or a visual signal such as a light, or a combination thereof. The amplified voice message is then furnished from the audio amplifier 108 to an audio loudspeaker within the alert 111 for message announcement and review by the user.

The decoder 107 may comprise a memory in which the received voice messages can be stored and recalled repeatedly for review by actuation of one or more controls 110.

In another aspect of the invention, portions of FIG. 1 can be equally interpreted as part of a dictation device, voice mail system, answering machine, or sound track editing device for example. By removing the wireless aspects of the system 100 including the removal of selective call transmitter 102 and radio frequency receiver 105, the system can be optionally hardwired from the voice compression circuitry 101 to the voice expansion circuitry 106 through the A/D 115 as shown with the dashed line. Thus, in a voice mail, answering machine, sound track editing or dictation system, an input device 114 would supply an acoustic input signal such as a speech signal to the terminal 113 having the voice compression circuitry 101. The voice expansion circuitry 106 and controls 110 would supply the means of listening and manipulating to the output speech signal in a voice mail, answering machine, dictation, sound track editing or other applicable system. This invention clearly contemplates that the time-scaling techniques of the claimed invention has many other applications besides paging. The paging example disclosed herein is merely illustrative of one of those applications.

Now referring to FIG. 2, there is shown a block diagram of a paging transmitter 102 and terminal 113 including an amplitude compression and filtering module 150 coupled to a time compression module 160 which is coupled to the selective call transmitter 102 and which transmits messages using aerial or antenna 103. Referring to FIGs . 3 and 4, a lower level block diagram of the block diagram of FIG. 2 is shown.

Please keep in mind that this compressed voice paging system is highly bandwidth efficient and intended to support typically 6 to 30 voice messages per 25 kHz channel using the basic concepts of quadrature amplitude (QAM) or single-side band (SSB) modulation and time scaling of speech signals. Preferably, in a first embodiment and also referring to FIG. 6, the compressed voice channel or voice communication resource consists of 3 sub-channels that are separated by 6250 Hz. Each sub-channel consists of two side-bands and a pilot carrier. Each of these two side-bands may have the same message in a first method or separate speech messages on each sideband or a single message split between the upper and lower sidebands in a second method. The single sub-channel has a bandwidth of substantially 6250 Hz with each side-band occupying a bandwidth of substantially 3125 Hz. The actual speech bandwidth is substantially 300-2800Hz. Alternatively, the quadrature amplitude modulation may be used where the two independent signals are transmitted directly via I and Q components of the signal to form each sub-channel signal. The bandwidth required for transmission is the same in the QAM and SSB cases.

Note that modules 150 and 160 in FIG. 2 can be repeated for use by each different voice signal (up to 6 times in 25 KHz wide channels and up to 14 times in 50KHz wide channels) to allow for the efficient and simultaneous transmission of (up to 6 in examples shown) voice messages. They can all then be summed at a summing device (not shown, but see FIG. 5) and preferably processed as a composite signal in 102. A separate signal (not shown) contains the FM modulation of the FLEX™ protocol (as will be described later) which may optionally be generated in software or as a hardware FM signal exciter.

Preferably, in the examples shown herein, an incoming speech message is received by the terminal 113. The present system preferably uses a time-scaling scheme or technique to achieve the required compression. The preferred compression technique used in the present invention requires certain parameters specific to the incoming message to provide an optimum quality. Preferably, the technique of time-scale compression processes the speech signal into a signal having the same bandwidth characteristics as uncompressed speech. (Once these parameters are computed, speech is compressed using the desired time-scaling compression technique). This time-scaled compressed speech is then encoded using a digital coder to reduce the number of bits required to be distributed to the transmitters. In the case of a paging system, the encoded speech distributed to the transmitters of multiple simulcasting sites in a simulcasting paging system would need to be decoded once again for further processing such as amplitude compression. Amplitude compression of the incoming speech signals (preferably using a syllabic compander) is used at the transmitter to give protection against channel impairments.

A time scaling technique known as Waveform Similarity based Overlap-Add technique or WSOLA encodes speech into an analog signal having the same bandwidth characteristics as uncompressed speech. This property of WSOLA allows it to be combined with SSB or QAM modulation such that the overall compression achieved is the product of the bandwidth compression ratio of multiple QAM or SSB subchannels (in our example, 6 voice channels) and the time compression ratio of WSOLA (typically between 1 and 5). In the present invention, a modified version of WSOLA, later described and referred to as "WSOLA-SD" is used. WSOLA-SD retains the compatibility characteristics of WSOLA that allows the combination with SSB or QAM modulation.

Preferably, an Adaptive Differential Pulse Coded Modulation coder (ADPCM) is used to encode the speech into data that is subsequently distributed to the transmitters. At the transmitter, the digital data is decoded to obtain WSOLA-SD compressed speech which is then amplitude companded to provide protection against channel noise. This signal is Hilbert transformed to obtain a single-sideband signal. Alternatively, the signal is quadrature modulated to obtain a QAM signal. A pilot carrier is then added to the signal and the final signal is interpolated, preferably, to a 16 kHz sampling rate and converted to analog. This is then modulated and transmitted.

The present invention can operate as a mixed-mode (voice or digital) one or two way communications system for delivering analog voice and/or digital messages to selective call receiver units on a forward channel (outbound from the base transmitter) and for

9

receiving acknowledgments from the same selective call receiver units which additionally have optional transmitters (on an optional reverse channel (inbound to a base receiver). The system of the present invention preferably utilizes a synchronous frame structure similar to FLEX™ (a high speed paging protocol by Motorola, Inc. and subject of U.S. Patent No. 5,282,205, which is hereby incorporated by reference) on the forward channel for both addressing and voice messaging. Two types of frames are used: control frames and voice frames. The control frames are preferably used for addressing and delivery of digital data to selective call receivers in the form of portable voice units (PVU's). The voice frames are used for delivering analog voice messages to the PVU's. Both types of frames are identical in length to standard FLEX™ frames and both frames begin with the standard FLEX™ synchronization. These two types of frames are time multiplexed on a single forward channel. The frame structure for the present invention will be discuss in greater detail later on with regard to FIGs 10, 11 and 12.

With regard to modulation, two types of modulation are preferably used on the forward channel of the present invention: Digital FM (2-level and 4-level FSK) and AM (SSB or QAM with pilot carrier). Digital FM modulation is used for the sync portions of both types of frames, and for the address and data fields of the control frames. AM modulation (each sideband maybe used independently or combined together in a single message) is used in the voice message field of the voice frames. The digital FM portions of the transmission support 6400 BPS (3200 Baud symbols) signaling. The AM portions of the transmissions support band limited voice (2800 Hz) and require 6.25 KHz for a pair of voice signals. The protocol, as will be shown later, takes advantage of the reduced AM bandwidth by subdividing a full channel into 6.25 KHz subchannels, and by using each subchannel and the AM sidebands for independent messages.

Voice System of the present invention is preferably designed to operate on either 25 KHz or 50 KHz forward channels, but other size spectrum is certainly within contemplation of the present invention. A 25 KHz forward channel supports a single FM control signal during control frames, and up to 3 AM subchannels (6 independent signals) during the message portion of voice frames. A 50 KHz forward channel supports two FM control signals operated in time lock during control frames, and up to 7 AM subchannels (14 independent signals) during the message portion of voice frames. Of course, other configurations using different size bandwidths and numbers of subchannels and signals are contemplated within the present invention. The examples disclosed herein are merely illustrative and indicative of the potential broad scope of the claims herein.

In addition to the spectrum efficiency achieved through modulation and sub-channelization of the spectrum, the present invention, in another embodiment, can utilize a speaker dependent voice compression technique that time scales the speech by a factor of 1 to 5 times. By using both AM sidebands (alternatively, the 2 QAM components) of a subchannel for different portions of the same message or different messages, the overall compression factor per subchannel is 2 to 10 times. Voice quality will typically decrease with an increasing time-compression factor. The compression technique preferably used in the voice system of the present invention is a modified form of a known time-scaling technique known as Waveform Similarity based Overlap-Add technique (WSOLA) as previously mentioned. The modified form of WSOLA is dependent upon the particular speaker or speech used, hence the name 'WSOLA-SD" for "WSOLA-Speaker dependent", which will be discussed later on.

Operation of the present invention is enhanced when a reverse (inbound to the base receiver) channel is available. The frequency division simplex mode of operation is one inbound operating mode supported. (U.S. Patent Nos. 4,875,038 and 4,882,579, both assigned to assignee of the present invention, Motorola, Inc., illustrate the use of multiple acknowledge signals on an inbound channel and are incorporated herein by reference). In frequency division simplex, a separate dedicated channel (usually paired with the outbound channel) is provided for inbound transmissions. Inbound data rates of 800 to 9600 BPS are contemplated within a channel bandwidth of 12.5 KHz.

The system of the present invention can be operated in one of several modes depending on the availability of a reverse channel. When no reverse channel is available, the system is preferably operated in simulcast mode for both addressing and voice messaging. When a reverse channel is provided, the system can be operated in a targeted message mode whereby the messages are broadcast only on a single or a subset of transmitters located near the portable voice unit. The targeted message mode is characterized by simulcast addressing to locate the portable voice unit, the portable voice unit's response on the reverse channel provides the location, followed by a localized message transmission to the portable voice unit. The targeted message mode of operation is advantageous in that it provides the opportunity for subchannel reuse; and consequently, this mode of operation can lead to increased system capacity in many large systems.

FIG. 3 illustrates a block diagram of a first embodiment of a transmitter 300 in accordance with the present invention. An analog speech signal is input to an anti-aliasing low pass filter 301 which strongly attenuates all frequencies above one-half the sampling rate of an analog-to-digital converter (ADC) 303 which is further coupled to the filter 301. The ADC 303 preferably converts the analog speech signal to a digital signal so that further signal processing can be done using digital processing techniques. Digital processing is the preferred method, but the same functions could also be performed with analog techniques or a combination of analog and digital techniques.

A band pass filter 305 coupled to the ADC 303 strongly attenuates frequencies below and above its cutoff frequencies. The lower cutoff frequency is preferably 300 Hz which allows the significant speech frequencies to pass, but attenuates lower frequencies which would interfere with a pilot carrier. The upper cutoff frequency is preferably 2800 Hz which allows the significant speech frequencies to pass but attenuates higher frequencies which would interfere with adjacent transmission channels. An automatic gain control (AGC) block 307 preferably coupled to the filter 305 equalizes the volume level of different voices.

12

A time compression block 309 preferably coupled to the AGC block 307 shortens the time required for transmission of the speech signal while maintaining essentially the same signal spectrum as at the output of the bandpass filter 305. The time compression method is preferably WSOLA-SD (as will be explained later on), but other methods could be used. An amplitude compression block 311 , and the corresponding amplitude expansion block 720 in a receiver 700 (FIG. 7), form a companding device which is well known to increase the apparent signal-to-noise ratio of the received speech. The companding ratio is preferably 2 to 1 in decibels, but other ratios could be used in accordance with the present invention. In the particular instance of a communication system such as a paging system, the devices 301-309 may be included in a paging terminal (113 of FIG. 1) and the remaining components in FIG. 3 could constitute a paging transmitter (102 of FIG. 1). In such a case, there would typically be a digital link between the paging terminal and paging transmitter. For instance, the signal after block 309 could be encoded using a pulse code modulation (PCM) technique and then subsequently decoded using PCM to reduce the number of bits transferred between the paging terminal and paging transmitter.

In any event, a second band pass filter 308 coupled to the amplitude compression block 311 strongly attenuates frequencies below and above its cutoff frequencies to remove any spurious frequency components generated by the AGC 307, the time compression block 309 or the amplitude compression block 311. The lower cutoff frequency is preferably 300 Hz which allows the significant speech frequencies to pass, but attenuates lower frequencies which would interfere with the pilot carrier. The upper cutoff frequency is preferably 2800 Hz which allows the significant speech frequencies to pass but attenuates higher frequencies which would interfere with adjacent transmission channels.

The time compressed speech samples are preferably stored in a buffer 313 until an entire speech message has been processed. This allows the time compressed speech message to then be transmitted as a whole. This buffering method is preferably used for paging service (which is typically a non real time service). Other buffering methods may be preferable for other applications. For example, for an application involving two-way real time conversation, the delay caused by this type of buffering could be intolerable. In that case it would be preferable to interleave small segments of several conversations. For example, if the time compression ratio is 3:1, then 3 real time speech signals could be transmitted via a single channel. The 3 transmissions could be interleaved on the channel in 150 millisecond bursts and the resulting delays would not be objectionable. The time compressed speech signal from the buffer 313 is applied to both to a Hilbert transform filter 323 and to a time delay block 315 which has the same delay as the Hilbert transform filter, but does not otherwise affect the signal.

The output of the time delay block 315 (through the summing circuit 317) and the Hilbert transform filter 323 form, respectively, the in-phase (I) and quadrature (Q) components of an upper sideband (USB) single sideband (SSB) signal. The output of the time delay and the negative (325) of the Hilbert transform filter form, respectively, the in-phase (I) and quadrature (Q) components of a lower sideband (LSB) single sideband signal. Thus the transmission may be on either the upper or lower sideband, as indicated by the dotted connection.

While the upper sideband is used to transmit one time compressed speech signal, the lower sideband can be used to simultaneously transmit a second time compressed speech signal by using another similar transmitter operating on the lower sideband. SSB is the preferred modulation method because of efficient use of transmission bandwidth and resistance to crosstalk. Double sideband Amplitude Modulation (AM) or frequency modulation (FM) could be used, but would require at least twice the bandwidth for transmission. It is also possible to transmit one time compressed speech signal directly via the I component and a second time compressed speech signal directly via the Q component, however, in the present embodiment this method is subject to crosstalk between the two signals when multipath reception occurs at the receiver.

A direct current (DC) signal is added to the I component of the signal to generate the pilot carrier, which is transmitted along with

14

the signal and used by the receiver (700) to substantially cancel the effects of gain and phase variations or fading in the transmission channel. The I and Q components of the signal are converted to analog form by digital-to-analog converters (DAC) 319 and 327 respectively. The two signals are then filtered by low pass reconstruction filters 321 and 329 respectively to remove spurious frequency components resulting from the digital-to-analog conversion process. A quadrature amplitude modulation (QAM) modulator 333 modulates the I and Q signals onto a radio frequency (RF) carrier at low power level. Other modulation methods, e.g. direct digital synthesis of the modulated signal would accomplish the same purpose as the DACs (319 and 327), reconstruction filters (321 and 329), and QAM modulator 333. Finally, a linear RF power amplifier 335 amplifies the modulated RF signal to the desired power level, typically 50 watts or more. Then, the output of the RF power amplifier 335 is coupled to the transmitting antenna. Other variations can produce essentially the same results. For example, the amplitude compression could be performed before the time compression, or omitted altogether and the device would still perform essentially the same function.

FIG. 4 illustrates a block diagram of a second embodiment of a transmitter 400 in accordance with the present invention. In FIG. 4, both the upper and lower sidebands are used to simultaneously transmit different portions of the same time compressed signal. The transmitter 400 preferably includes an anti-alias filter 404, an ADC 403, a bandpass filter 405, an AGC 407, a time compression block 409, an amplitude compression block 411, and a bandpass filter 408 coupled and configured as in FIG. 3. Operation of the transmitter of FIG. 4 is the same as in FIG. 3 until an entire speech message has been processed and stored in a buffer 413. The time compressed speech samples stored in the buffer 413 are then divided to be transmitted on either the upper or lower sideband. Preferably, the first half of the time compressed speech message is transmitted via one sideband and the second half of the time compressed speech message is transmitted via the other sideband (or alternatively on each of the I and Q components directly).

The first portion of time compressed speech signal from the buffer 413 is applied to both a first Hilbert transform filter 423 and to a first time delay block 415 which has the same delay as the Hilbert transform filter 423 but does not otherwise affect the signal. The output of the first time delay (through summing circuit 417) and the first Hilbert transform filter 423 (through summing circuit 465) are In-Phase (I) and Quadrature Phase (Q) signal components which, when coupled to I and Q inputs of the QAM modulator, generate upper sideband signal having information only from the first portion of time compressed speech samples. The second time compressed speech signal from the buffer 413 is applied to both a second Hilbert transform filter 461 and to a second time delay block 457 which has the same delay as the Hilbert transform filter 461 but does not otherwise affect the signal. The output of the second time delay (through summing circuits 459 and 417) and the negative (463) of the output of the second Hilbert transform filter 461 (and again, through summing circuit 465) are In-Phase (I) and Quadrature Phase (Q) signal components which, when coupled to I and Q inputs of the QAM modulator, generate upper sideband signal having information only from the second portion of time compressed speech samples. The I components of the upper and lower sideband signals are added with a DC pilot carrier component (through summing circuit 459) to form a composite I component for transmission. The Q components of the upper and lower sideband signals are added (through summing circuit 465) to form a composite Q component for transmission. It will be appreciated that elements 415, 423, 457, 461, 417, 459, 463, 465, 419, 427, 421, and 429 form a preprocessor which generates preprocessed I and Q signal components, which when coupled to the QAM modulator 453 generate the low level subchannel signal with a subcarrier F_A, having two single sideband signals. which have independent information on each sideband.

The transmitter 400 further comprises DACs 419 and 427, reconstruction filters 421 and 429, QAM modulator 433, and RF power amplifier 455 arranged and constructed as described in FIG. 3. Operation of the rest of the transmitter of FIG. 4 is the same as in FIG. 3.

16

Preferably, in both transmitters 300 and 400 of FIGs. 3 and 4 respectively, only the anti-alias filters, the reconstruction filters, the RF power amplifier and optionally the Analog to Digital converter and digital to analog converters are separate hardware components. The remainder of the devices can preferably be incorporated into software which could be run on a processor, preferably a digital signal processor.

FIG. 7 illustrates a block diagram of a receiver 700 which preferably operates in conjunction with the transmitter 300 of FIG. 3 in accordance with the present invention. A receiving antenna is coupled to a receiver module 702. The receiver module 702 includes conventional receiver elements, such as RF amplifier, mixer, bandpass filter, and intermediate frequency (IF) amplifier (not shown). A QAM demodulator 704 detects the I and Q components of the received signal. An analog-to-digital converter (ADC) 706 converts the I and Q components to digital form for further processing. Digital processing is the preferred method, but the same functions could also be performed with analog techniques or a combination of analog and digital techniques. Other methods of demodulation, e.g., a sigma-delta converter, or direct digital demodulation, would accomplish the same purpose as the QAM demodulator 704 and ADC 706.

A feedforward automatic gain control (AGC) block 708 uses the pilot carrier, transmitted along with the time compressed speech signal, as a phase and amplitude reference signal to substantially cancel the effects of amplitude and phase distortions occurring in the transmission channel. The outputs of the feedforward automatic gain control are corrected I and Q components of the received signal. The corrected Q component is applied to a Hilbert transform filter 712, and the corrected I component is applied to a time delay block 710 which has the same delay as the Hilbert transform filter 712 but does not otherwise affect the signal.

If the time compressed speech signal was transmitted on the upper sideband, the output of the Hilbert transform filter 712 is added (through summing circuit 714) to the output of the time delay block 710 to produce the recovered time compressed speech signal. If the time compressed speech signal was transmitted on the lower

17

sideband, the output of the Hilbert transform filter 712 is subtracted (716) from the output of the time delay block 710 to produce the recovered time compressed speech signal. The recovered time compressed speech signal is preferably stored in a buffer 718 until an entire message has been received. Other buffering methods are also possible. (See the discussion with FIG. 3.)

An amplitude expansion block 720 works in conjunction with the amplitude compression block 311 of FIG. 3 to perform the companding function. A time expansion block 722 works in conjunction with the time compression block 309 of FIG. 3 and preferably reconstructs the speech into its natural time frame for an audio output through transducer 724 or other time frames as other applications may suggest. One application could optionally include the transfer of digitized voice to a computing device 726, where the receiver-to-computer interface can be a PCMCIA or RS-232 interface or any number of interfaces known in the art. The time compression method is preferably WSOLA-SD, but other methods could be used, so long as complementary methods are used in the transmitter and receiver. Other variations in configuration can produce essentially the same results. For example, the amplitude compression could be performed after the time compression, or omitted altogether and the device would still perform essentially the same function.

FIG. 8 illustrates a block diagram of a receiver 750 which operates in conjunction with the transmitter of FIG. 400 in accordance with the present invention. The receiver of FIG. 8 comprises an antenna, receiver module 752, a QAM modulator 754, an ADC 756, a Feed-forward AGC 758, a time delay block 760, and a Hilbert transform filter 762 arranged and constructed as described in FIG. 7. Operation of the receiver of FIG. 8 is the same as FIG. 7, up to the output of the time delay block 760 and Hilbert transform filter 762. The output of the Hilbert transform filter 762 is added to the output of the time delay block 760 (through summing circuit 764) to produce the recovered time compressed speech signal corresponding to the first half of the speech message which was transmitted on the upper sideband. The output of the Hilbert transform filter 762 is subtracted (766) from the output of the time delay block 760 to produce the recovered time compressed speech signal corresponding to the second half of the speech message which was transmitted on the lower sideband.

The two recovered time compressed speech signals are stored in either respective upper sideband and lower sideband buffers 768 or 769 until the entire message has been received. Then, the signal corresponding to the first half of the message and the signal corresponding to the second half of the message are applied sequentially to the amplitude expansion block 770. An amplitude expansion block 770 works in conjunction with the amplitude compression block 411 of FIG. 4 to perform the companding function.

The operation of the rest of the receiver of FIG. 8 is the same as FIG. 7. A time expansion block 772 works in conjunction with the time compression block 409 of FIG. 4 and preferably reconstructs the speech into its natural time frame or other time frames as other applications may suggest or require. The time compression method is preferably WSOLA-SD, but other methods could be used, so long as complementary methods are used in the transmitter and receiver. Other configurations can produce essentially the same results. For example, the amplitude compression could be performed after the time compression, or omitted altogether and the device would still perform essentially the same function.

As with the implementation of the transmitters of FIGs. 3 and 4 , many of the components in FIGs. 7 and 8 can be implemented in software including, but not limited to the AGCs, the single-sideband or QAM demodulators, summation circuits, the amplitude expansion blocks, and the time expansion blocks. All the other components are preferably implemented in hardware.

If the speech processing, encoding and modulation portion of the present invention were to be implemented into hardware, the implementation of FIG. 5 could be used. For instance, transmitter 500 of FIG. 5 would include a series of pairs of single-sideband exciters (571-576) set to the frequencies of their respective pilot carriers (581-583). Exciters 571-576 and pilot carriers 581-583 correspond to the separate voice processing paths. All these signals, including a signal from an FM signal exciter 577 (for the digital FM modulation used for the synchronization, address and data fields previously described) would be fed into a summing amplifier 570 which in turn is amplified by a linear amplifier 580 and subsequently transmitted. The low level output of FM exciter 577 is also linearly combined in summing amplifier 570. The composite output signal of summing amplifier 570 is amplified to the desired power level, usually 50 watts or more, by linear RF power amplifier 580. The output of linear RF power amplifier 580 is then coupled to the transmitting antenna.

Other means could be used to combine several subchannel signals. For example, the several digital baseband I and Q signals, obtained at the outputs of 417 and 465 in Fig. 4, could be translated in frequency to their respective subcarrier offset frequencies, combined in digital form, then converted to analog form for modulation onto the carrier frequency.

Referring to FIG. 9, there is shown another receiver unit 900 in accordance with the present invention. Receiver 900 additionally incorporates a means for detecting and decoding the FM modulated control signals that are used in the FLEX™ signaling protocol. Block 902 is the receiver front end and an FM back end. A digital automatic frequency controller (DAFC) and automatic gain controller (AGC) are incorporated into block 902. Block 906 includes the radio processor with a support chip 950 and Blocks 911, 914, and 916 include all the output devices. Block 904 is the battery saver or battery economy circuit which operates under control of the processor 906. Block 850 is the linear decoder followed by an analog-to-digital converter and random access memory (RAM) Block 868. The receiver Block 902 is preferably a modified FM receiver including the addition of a DAFC as described in U.S. Patent No. 5,239,306 (which is assigned to the assignee of the present invention and which is hereby incorporated by reference herein), an AGC, and which provides for an intermediate frequency (IF) output at a point following most of the receiver gain but prior to the FM demodulator.

The same processor that controls Motorola's FLEX™ protocol compatible pagers would adequately handle all the protocol functions in the present invention including the address recognition and message decoding of an FM demodulated signal. Additionally, in response to an FM modulated address (and perhaps message pointer code words), the processor 906 initiates the operation of the analog-to-digital conversion and of the RAM Block 868. Block 868 samples either or both the I (In-phase) and Q (quadrature) linearly modulated signals at the outputs of the linear decoder block 850. The signal samples are written directly to RAM with the aid of an address counter and in response to a control signal from the processor 906.

A voice can be sent as an SSB signal occupying a single voice bandwidth on the channel, or equivalently on either of the I or Q channels as was described earlier. Each of the I and Q signals simultaneously occupy the same RF bandwidth as two analog-single sidebands (SSB). Voice bandwidths are on the order of 2.8KHz, so a typical signal sampling rate of about 6.4 KHz each is required of the analog-to-digital converter if analog-SSB is recovered from the I and Q channel information. The analog-to-digital converter samples with 8 bit precision (although as much as 10 bits is preferred). Direct memory access by the analog-to-digital converter allows the use of a processor whose speed and power are not a direct function of the channel data rate. That is, a microprocessor can be used with direct memory access, whereas, a significantly higher speed processor would be required if the analog-to-digital converted data were read to memory through the microprocessor.

The analog-to-digital converter (A/D), the dual port RAM and the address counter are grouped as block 868. A second RAM I/O port can be serial or parallel, and operates at a 6 or 12 K sample per second rate. A second RAM I/O port is provided so that the processor can extract the sampled voice or data, process the demodulation function, and expand the compressed voice or format the data. The restored voice is played back through the voice processor 914 and transducer 916, while formatted data can be displayed on display 911.

Again, referring to FIG. 9, an expanded electrical block diagram is used to describe in further detail the receiver operation of the dual mode communication receiver of the present invention. The transmitted information signal, modulated in the FM modulation format, or in a linear modulation format (such as SSB), is intercepted by the antenna 802 which couples the information signal to the receiver section 902, and in particular to the input of the radio frequency (RF) amplifier 806. The message information is transmitted on any suitable RF channel, such as those in the VHF bands and UHF bands. The RF amplifier 806 amplifies the received information signal, such as that of a signal received on a 930 MHz paging channel frequency, coupling the amplified information signal to the input of the first mixer 808. The first oscillator signal, which is generated in the preferred embodiment of the present invention by a frequency synthesizer or local oscillator 810, also couples the first mixer 808. The first mixer 808 mixes the amplified information signal and the first oscillator signal to provide a first intermediate frequency, or IF, signal, such as a 45 MHz IF signal, which is coupled to the input of the first IF filter 812. It will be appreciated that other IF frequencies can be utilized as well, especially when other paging channel frequencies are utilized. The output of the IF filter 812 which is the on-channel information signal, is coupled to the input of the second conversion section 814, which will be described in further detail below. The second conversion section 814 mixes the on-channel information signal to a lower intermediate frequency, such as 455KHz, using a second oscillator signal, which is also generated by the synthesizer 810. The second conversion section 814 amplifies the resultant intermediate frequency signal, to provide a second IF signal which is suitable to be coupled to either the FM demodulator section 908 or to the linear output section 824.

Receiver section 804 operates in a manner similar to a conventional FM receiver, however, unlike a convention FM receiver, the receiver section 804 of the present invention also includes an automatic frequency control section 816 which is coupled to the second conversion section 814, and which appropriately samples the second IF signal to provide a frequency correction signal which is coupled to the frequency synthesizer 810 to maintain the receiver tuning to the assigned channel. The maintenance of receiver tuning is especially important for the proper reception of QAM (that is, I and Q components) and/or SSB information which is transmitted in the linear modulation format. The use of a frequency synthesizer to generate the first and second oscillator frequencies enables the operation selection of the receiver on multiple operating frequencies, selected such as by code memory programming and/or by parameters received over the air, as for example, in the FLEX™ protocol. It will be appreciated that other oscillator circuits, such as fixed frequency oscillator circuits which can be adjusted by a frequency correction signal from the automatic frequency control section 816, can be utilized as well.

An automatic gain control 820 is also coupled to the second conversion section 814 of the dual mode receiver of the present invention. The automatic gain control 820 estimates the energy of samples of the second IF signal and provides a gain correction signal which is coupled to the RF amplifier 806 to maintain a predetermined gain for the RF amplifier 806. The gain correction signal also couples the second conversion section 814 to maintain a predetermined gain for the second conversion section 814. The maintenance of the gain of the RF amplifier 806 and the second conversion section 814 is required for proper reception of the high speed data information transmitted in the linear modulation format, and further distinguishes the dual mode receiver of the present invention from a conventional FM receiver.

When the message information or control data is transmitted in the FM modulation format, the second IF signal is coupled to the FM demodulator section 908, as will be explained in detail below. The FM demodulator section 908 demodulates the second IF signal in a manner well known to one of skill in the art, to provide a recovered data signal, which is a stream of binary information corresponding to the received address and message information transmitted in the FM modulation format. The recovered data signal coupled to the input of a microcomputer 906, which function as a decoder and controller, through an input of input/output port, or I/O port 828. The microcomputer 906 provide complete operational control of the communication receiver 900, providing such functions as decoding, message storage and retrieval, display control, and alerting, just to name a few. The device 906 is preferably a single chip microcomputer such as the MC68HC05 microcomputer manufactured by Motorola, and includes CPU 840 for operational control. An internal bus 830 connects each of the operational elements of the device 906. I/O port 828 (shown split in FIG. 9) provides a plurality of control and data lines providing communications to device 906 from external circuits, such as the battery saver switch 904, audio processor 914, a display 911, and digital storage 868. A timing means, such as timer 834 is used to generate the timing signals required for the operation of the communication receiver, such as for battery saver timing, alert timing, and message storage and display timing. Oscillator 832 provides the clock for operation of CPU 840, and provides the reference clock for timer 834. RAM 838 is used to store information utilized in executing the various firmware routines controlling the operation of the communication receiver 900, and can also be used to store short messages, such as numeric messages. ROM 836 contains the firmware routines used to control the device 906 operation, including such routines as required for decoding the recovered data signal, battery saver control, message storage and retrieval in the digital storage section 868, and general control of the pager operation and message presentation. An alert generator 842 provides an alerting signal in response to decoding the FM modulated signaling information. A code memory 910 (not shown) couples the microcomputer 906 through the I/O port 828. The code memory is preferably an EEPROM (electrically erasable programmable read only memory) which stores one or more predetermined addresses to which communication receiver 900 is responsive.

When the FM modulated signaling information is received, it is decoded by the device 906, functioning as a decoder in a manner well known to one skilled in the art. When the information in the recovered data signal matches any of the stored predetermined addresses, the subsequently received information is decoded to determine if additional information is directed to the receiver which is modulated in the FM modulation format, or if the additional information is modulated in the linear modulation format. When the additional information is transmitted in the FM modulation format, the recovered message information is received and stored in the microcomputer RAM 838, or in the digital storage section 868, as will be explained further below, and an alerting signal is generated to alert generator 842. The alerting signal is coupled to the audio processing circuit 914 which drives transducer 916, delivering an audible alert. Other forms of sensible alerting, such as tactile or vibrating alert, can also be provided to alert the user as well.

When additional information is to be transmitted in the linear modulation format (such as SSB or "I and Q"), the microcomputer 906 decodes pointer information. The pointer information includes information indicating to the receiver on what combination of sidebands (or on what combination of I and Q components) within the channel bandwidth that the additional information is to be transmitted. The device 906 maintains the operation of monitoring and decoding information transmitted in the FM modulation format, until the end of the current batch, at which time the supply of power is suspended to the receiver until the next assigned batch, or until the batch identified by the pointer is reached, during which high speed data is transmitted. The device 906, through I/O port 828 generates a battery saving control signal which couples to battery saver switch 904 to suspend the supply of power to the FM demodulator 908, and to supply power to linear output section 824, the linear demodulator 850, and the digital storage section 868, as will be described below.

The second IF output signal, which now carries the SSB (or "I and Q") information is coupled to the linear output section 824. The output of the linear output section 824 is coupled to the quadrature detector 850, specifically to the input of the third mixer 852. A third local oscillator also couples to the third mixer 852, which is preferably in the range of frequencies from 35-150kHz, although it will be appreciated that other frequencies may be utilized as well. The signal from the linear output section 824 is mixed with the third local oscillator signal 854, producing a third IF signal at the output of the third mixer 852, which is coupled to a third IF amplifier 856. The third IF amplifier is a low gain amplifier which buffers the output signal from the input signal. The third output signal is coupled to an I channel mixer 858 and a Q channel mixer 860. The I/Q oscillator 862 provides quadrature oscillator signals at the third IF frequency which are mixed with the third output signals in the I channel mixer 858 and the Q channel mixer 860, to provide baseband I channel signals and Q channel signals at the mixer outputs. The baseband I channel signal is coupled to a low pass filter 864, and the baseband Q channel signal is coupled to a low pass filter 866, to provide a pair of baseband audio signals which represent the compressed and companded voice signals .

The audio signals are coupled to the digital storage section 868, in particular to the inputs of an analog to digital converter 870. The A/D converter 870 samples the signals at a rate at least twice the highest frequency component at the output of 864 and 866. The sampling rate is preferably 6.4 kilohertz per I and Q channel. It will be appreciated, that the data sampling rate indicated is for example only, and other sampling rates may be used depending upon the bandwidth of the audio message received.

During the batch when the high speed data is transmitted, the microprocessor 906 provides a count enabling signal which is coupled to the address counter 872. the A/D converter 870 is also enable to allow sampling of the information symbol pairs. The A/D converter 870 generates high speed sample clock signals which are used to clock the address counter 872 which in turn sequentially generates addresses for loading the sampled voice signals into a dual port random access memory 874 through data lines going from the converter 870 to the RAM 874. The voice signals which have been loaded at high speed into the dual port RAM 874 in real time, are processed by the microcomputer 906 after all voice signals have been received, thereby producing a significant reduction in the energy consumed by not requiring the microcomputer 906 to process the information in real time. The microcomputer 906 accesses the stored signals through data lines and address lines, and in the preferred embodiment of the present invention, processes the information symbol pairs to generate either ASCII encoded information in the case of alphanumeric data having been transmitted, or digitized sampled data in the case voice was transmitted. The digitized voice samples can alternatively stored in other formats such as BCD, CVSD, or LPC based forms and other types as required. In the case of time compressed voice signals, the I and Q components sampled by ADC converter 870 are further processed by CPU 840 via dual port RAM 874 and I/O 828 to (1) amplitude expand the audio signal and (2) time-expand the signal as was described in the similar operation of the receivers of FIGs. 7 and 8. The voice is then stored again in RAM 874. The ASCII encoded or voice data is stored in the dual port RAM until the information is requested for presentation by the communication receiver user. The stored ASCII encoded data is recovered by the user using switches (not shown) to select and read the stored messages. When the stored ASCII encoded message is to be read, the user selects the message to be read and actuates a read switch which enable microcomputer 906 to recover the data, and to present the recovered data to a display 911, such as a liquid crystal display. When a voice message is to be read, the user selects the message to be read and actuates a read switch which enables the microcomputer 906 to recover the data from the dual port RAM, and to present the recovered data to the audio processor 914 which converts the digital voice information into an analog voice signal which is coupled to a speaker 916 for presentation of the voice message to the user. The microcomputer 906 can also generate a frequency selection signal which is coupled to frequency synthesizer 810 to enable the selection of different frequencies as previously described.

Referring to FIG. 10, a timing diagram is shown which illustrates features of the FLEX™ coding format on outbound signaling utilized by the radio communication system 100 of FIG. 1, and which includes details of a control frame 330, in accordance with the preferred embodiment of the present invention. Control frames are also classified as digital frames. The signaling protocol is subdivided into protocol divisions, which are an hour 310, a cycle 320, frames 330, 430 a block 340, and a word 350. Up to fifteen 4 minute uniquely identified cycles are transmitted in each hour 310. Normally, all fifteen cycles 320 are transmitted each hour. Up to one hundred twenty eight 1.875 second uniquely identified frames including digital frames 330 and analog frames 430 are transmitted in each of the cycles 320. Normally, all one hundred twenty eight frames are transmitted. One synchronization and Frame Information signal 331 lasting one hundred fifteen milliseconds and 11 one hundred sixty millisecond uniquely identified blocks 340 are transmitted in each of the control frames 330. Bit rates of 3200 bits per second (bps) or 6400 bps are preferably used during each control frame 330. The bit rate during each control frame 330 is communicated to the selective call radios 106 during the synchronization signal 331. When the bit rate is 3200 bps, 16 uniquely identified 32 bit words are included in each block 340, as shown in FIG. 10. When the bit rate is 6400 bps 32 uniquely identified 32 bit words are included in each block 340 (not shown). In each word, at least 11 bits are used for error detection and correction, and 21 bits or less are used for information, in a manner well known to one of ordinary skill in the art. The bits and words 350 in each block 340 are transmitted in an interleaved fashion using techniques well known to one of ordinary skill in the art to improve the error correction capability of the protocol.

Information is included in each control frame 330 in information fields, comprising Frame structure information in a block information field (BI) 332, one or more selective call addresses in an address field (AF) 333, and one or more vectors in a vector field (VF) 334. The vector field 334 starts at a vector boundary 334. Each vector in the vector field 334 corresponds to one of the addresses in the address field 333. The boundaries of the information fields 332, 333, 334 are defined by block information field 332. Information fields 332, 333, 334 are variable, depending on factors such as the type of system information included in the sync and frame information field 331 and the number of addresses included in the address field 333, and the number and type of vectors included in the vector field 334.

Referring to FIG. 11, a timing diagram is shown which illustrates features of the transmission format of the outbound signaling protocol utilized by the radio communication system of FIG. 1, and which includes details of a voice frame 430, in accordance with the preferred embodiment of the present invention. Voice frames are also classified herein as analog frames. The durations of the protocol divisions hour 310, cycle 320, and frame 330, 430 are identical to those described with respect to a control frame in FIG. 10. Each analog frame 430 has a header portion 435 and an analog portion 440. The information in the synchronization and frame information signal 331 is the same as the synchronization signal 331 in a control frame 330. As described above, the header portion 435 is frequency modulated and the analog portion 440 of the frame 430, is amplitude modulated. A transition portion 444 exists between the header portion 435 and analog portion 440. In accordance with the preferred embodiment of the present invention, the transition portion includes amplitude modulated pilot subcarriers for up to three subchannels 441, 442, 443. The analog portion 440 illustrates the three subchannels 441, 442, 443 which are transmitted simultaneously, and each subchannel includes an upper sideband signal 401 and a lower sideband signal 402 (or alternatively, an in-phase and a quadrature signal). In the example illustrated in FIG. 11, the upper sideband signal 401 includes one message fragment 415, which is a first fragment of a first analog message. Included in the lower sideband 402 are four quality assessment signals 420, 422, 424, 426, four message segments 410, 412, 416, 418, and one segment 414 (unused in this example). The two segments 410, 412 are segments of a second fragment of the first analog message. The two segments 416, 418 are segments of a first fragment of a second analog message. The first and second analog messages are compressed voice signals which have been fragmented for inclusion in the first subchannel 441 of frame one 430 of cycle 2 of 320. The second fragment of the first message and the first fragment of the second message are each split to include a quality assessment signal 420, 426, which are repeated at predetermined positions in the lower sideband 402 of each of the three subchannels 441, 442, 443. The smallest segment of message included in an analog frame is defined as a voice increment 450, of which 88 are uniquely identified in each analog portion 440 of an analog frame 430. The quality assessments signals are preferably transmitted as unmodulated subcarrier pilot signals, are preferably one voice increment in duration, and preferably have a separation of no more than 420 milliseconds within an analog portion of a frame. It will be appreciated that more than one message fragment could occur between two quality assessment signals, and that message fragments are typically of varying integral lengths of voice increments.

29

Referring to FIG. 12, a timing diagram illustrating a control frame 330 and two analog frames of the outbound signaling protocol utilized by the radio communication system of FIG. 1 is shown, in accordance with the preferred embodiment of the present invention. The diagram of FIG. 12 shows an example of a frame zero (FIG. 10) which is a control frame 330. Four addresses 510, 511, 512, 513 and four vectors 520, 521, 522, 523 are illustrated. Two addresses 510, 511 include one selective call radio 106 address, while the other two addresses 512, 513 are for a second and third selective call radio 106. Each address 510, 511, 512, 513 is uniquely associated with one of the vectors 520, 521, 522, and 523 by inclusion of a pointer within each address which indicates the protocol position of (i.e., where the vector starts and how long it is) the associated vector.

In the example shown in FIG. 12, vectors 520, 521, 522, 523 are also uniquely associated with a message portion in one of the subchannels. Specifically, vector 520 can point to an upper sideband of subchannel 441 (see FIG. 11) and vector 522 can point to a lower sideband of subchannel 441. Similarly, vector 521 can point to both sidebands of subchannel 442. That is, in the case of subchannel 441, the example can show that two different message portions are carried by the upper and lower sidebands. In the case of subchannel 442, two halves of one message portion are carried by the upper and lower sidebands respectively. Thus, the vectors preferably include information therein to indicate which subchannel (i.e., which radio frequency) the receiver should look for a message, and also information to indicate whether two separate messages are to be recovered from the subchannel, or whether first and second halves of a single message are to be recovered.

One use for the embodiment where two different messages are simultaneously transmitted over upper and lower sidebands (or I and Q channels), respectively, is where one message is a direct voice paging message, and the other is a voice mailbox message, which is to be stored in the pager.

In accordance with the preferred embodiment of the present invention the vector position is provided by identifying the number of words 350 after the vector boundary 335 at which the vector starts,

30

and the length of the vector, in words. It will be appreciated that the relative positions of the addresses and vectors are independent for each other. The relationships are illustrated by the arrows. Each vector 520, 521, 522, 523 is uniquely associated with a message fragment 550, 551, 552, 553 by inclusion of a pointer within each vector which indicates the protocol position of (i.e., where the fragment starts and how long it is) the associated vector. In accordance with the preferred embodiment of the present invention the message fragment position is provided by identifying the frame 430 number (from 1 to 127), the subchannel 441, 442, 443 number (from one to three), the sideband 401, 402, (or I or Q) and the voice increment 450 where the message fragment starts, and the length of the message fragment, in terms of voice increments 450. For example, vector three 522 includes information which indicates that message two, fragment one 552, which is intended for selective call transceiver 106 having selective call address 512, is located starting at voice increment forty six 450 (the voice increments 450 are not identified in FIG. 12) of frame one 560, and vector thirteen 523 includes information which indicates that message nine fragment one 553, which is intended for selective call transceiver 106 having selective call address 513, is located starting at voice increment zero 450 (the voice increments 450 are not shown in FIG. 12) of frame five 561 .

It will be appreciated that, while voice signals are described in accordance with the preferred embodiment of the present invention, other analog signals, such as modem signals or dual tone multi-frequency (DTMF) signals, can alternatively be accommodated by the present invention. It should also be appreciated that the block information used in the frame structure previously described can be used to implement further enhancements that would allow for greater overall throughput in a communication system and allow for additional features. For instance, a message sent to a portable voice unit can request that an acknowledgment signal sent back to the system include information that would identify the transmitter it was receiving its messages from. Thus, frequency reuse in a simulcast system can be achieved in this way by transmitting messages to the given portable voice unit using the one transmitter

31

required to reach the portable voice unit. Additionally, once the system knows the location of the portable voice unit, implementing target messaging logically follows.

In another aspect of the present invention, the time-scaling technique, previously described as WSOLA has some existing disadvantages when used in conjunction with the present invention. Thus, a technique was developed that modifies WSOLA to become speaker dependent and appropriately named "WSOLA-SD". To further understand our modification of WSOLA to form WSOLA-SD, a brief description of WSOLA follows.

A technique called Waveform similarity based Overlap-Add technique (WSOLA) can achieve high-quality time-scale modification compared to other techniques and is also much simpler than other methods. When used to speed up or slow down speech, the quality of speech is not very good even with the WSOLA technique. The reconstructed speech contains a lot of artifacts like echoes, metallic sounds and reverberations in the background. This aspect of the present invention describes several enhancements to overcome this problem and minimize the artifacts present. Many parameters in the WSOLA algorithm have to be optimized to achieve the best quality possible for a given speaker and required compression/expansion or time-scaling factor. This aspect of the invention deals with determining those parameters and how to incorporate them in compression/expansion or time-scaling of speech signals with improvement in the quality of the recovered speech or voice signal.

The WSOLA Algorithm: Let $x(n)$ be the input speech signal to be modified, $y(n)$ the time-scale modified signal and α be the time-scaling parameter. If α is less than 1 then the speech signal is expanded in time. If α is greater than 1 then the speech signal is compressed in time.

Referring to FIGs. 13-17, timing diagrams for several iterations of the WSOLA time-scaling (compression) method is shown for comparison to the preferred method of WSOLA-SD of the present invention. Assuming that the input speech signals are appropriately digitized and stored, FIG. 13 illustrates the first iteration of the WSOLA method on an uncompressed speech input signal. The

WSOLA method requires a time scale factor of α (which we assume is equal to 2 for this example, where if $\alpha>1$ we have compression and if $\alpha<1$ we have expansion) and an arbitrary analysis segment size (Ss) which is independent of the input speech characteristics, and in particular, independent of pitch. An overlap segment size So is computed as 0.5*Ss and is fixed in WSOLA. The first Ss samples are copied directly to the output as shown in FIG. 14. Let the index of the last sample in the output be I_{f1}. An overlap index O_1 is determined as Ss/2 samples from the end of the last available sample in the output. Now the samples which would be overlap added are between O_1 and I_{f1}. Search index (S_1) is determined as $\alpha{*}O_1$. After an initial portion of the input signal is copied into the output, a determination is made of the moving window of samples from the input. The window is determined around the search index S_1. Let the beginning of the window be S_i - L_{offset} and the end be $S_i + H_{offset}$. In the first iteration, i = 1. Within the window, the best correlating So samples are determined using a Normalized Cross-Correlation equation given by:

$$R(k)=\frac{\sum_{j=0}^{j=S_o}x(s_i+k+j)y(o_i+j)}{\left[\sum_{j=0}^{j=S_o}x^2(s_i+k+j)\sum_{j=0}^{j=S_o}y^2(o_i+j)\right]^{1/2}}$$

where

$$k=S_i-L_{offset},S_i+H_{offset}$$

The lag $k=m$ for which the normalized $R(k)$ is maximum is determined. The best index Bi is given by $Si+m$. Note that other schemes like Average Magnitude Difference Function (AMDF) and other correlation functions can be used to find the best matching waveform. The So samples beginning at B1 are then multiplied by an increasing ramp function (although other weighting functions could be used) and added to the last So samples in the output. Prior to the addition, the So samples in the output are multiplied by a decreasing ramp function (although other weighting functions could be used here as well). The resulting samples of the addition will replace the last So samples in the input. Finally, the next So samples which immediately follow the prior best matching So samples are then copied to the end of the output for use in the next iteration. This would be the end of the first iteration in WSOLA.

Referring to FIGs. 15 and 16 for the next iteration, we need to compute a new overlap index O_2, similarly to O_1. Likewise, a new search index S_2 and corresponding search window is determined as was done in the previous iteration. Once again, within the search window, the best correlating So samples are determined using the cross-correlation equation previously described above, where the beginning of the best samples determined is B_2. The So samples beginning at B2 are then multiplied by an increasing ramp function and added to the last So samples in the output. Prior to the addition, the So samples in the output are multiplied by a decreasing ramp function. The resulting samples of the addition will replace the last So samples in the input. Finally, the next So samples which immediately follow the prior best matching So samples are then copied to the end of the output for use in the next iteration, where future i^{th} iterations would have an overlap index O_i, a Search index S_i, last sample in output I_{fi}, and a best index B_i.

FIG. 17 shows the resultant output from the previous two iterations described with reference to FIGs. 13-16. Once should note that there is no overlap in the resultant output signal between the two iterations. If the method were to continue in a similar fashion, the WSOLA method would time scale (compress) the entire speech signal, but there would never be any overlap between the results of each of the iterations. WSOLA time-scale expansion is done in a similar fashion.

Several drawbacks or disadvantages of WSOLA with respect to the preferred method of the present invention (WSOLA-SD) become apparent. These drawbacks should be kept in mind as you follow the next examples of the WSOLA-SD method shown in FIGs. 18-23. A primary drawback of WSOLA includes the inability to obtain the optimum quality of time scaled speech because a fixed analysis segment size (Ss) is used for all input speech irrespective of the pitch characteristics. For instance, if the Ss was too large for the input speech signal, the resultant speech upon expansion would

34

include echoes and reverberations. Further, if the Ss is too small for the input speech signal, then the resultant speech upon expansion would sound raspy.

A second significant drawback of WSOLA results when compression rates (α) are greater than 2. In such instances, the separation of the moving window between iterations may cause the method to skip significant input speech components, thereby seriously affecting the intelligibility of the resultant output speech. Increasing the size of the moving windows to compensate for the non-overlapping search windows during iterations causes further skipping of some input speech as a result of the cross-correlation function and further causes variable time-scaling that noticeably affects the resultant output speech.

A third drawback of the WSOLA method involves its failure to provide a designer or user the flexibility (for a given time-scaling factor (α)) with respect to quality of speech and complexity of computation for a given system having given restraints. This is particularly apparent because the degree of overlap (f) is fixed at 0.5 in the WSOLA method. Thus, in an application that requires high quality speech reproduction, assuming adequate processing power and memory, the WSOLA-SD method of the present invention can use a higher degree of overlap at the expense of added computational complexity to provide higher quality speech reproduction. On the other hand, in an application that is limited by processing power, memory or other constraints, the degree of overlap can be lowered in WSOLA-SD so that the quality of speech is sacrificed only to the extent desired, taking into account the particular application constraints at hand.

FIG. 25 illustrates an overall block diagram of WSOLA-SD method. In this block diagram *Ss*, *f* and α are computed depending on whether we are compressing or expanding speech. This WSOLA-SD algorithm provides great improvement in the quality of reconstructed speech over WSOLA alone. The WSOLA-SD method is speaker dependent, particularly to the pitch of a particular speaker. Thus, a pitch determination 12 is done before an analysis segment sized is determined (14). For a given f and α (which can be modified dependent upon the pitch determination 12, providing a modified alpha (16)), WSOLA-SD time scales (18) the speech. The time-scaling can either be expansion or compression of the input signal. Alternatively, frequency-scaled signal can be obtained by interpolating the time-scaled signal by a factor of α if $\alpha > 1$ or by decimating the time-scaled signal by a factor of $1/\alpha$ if $\alpha < 1$. Interpolation and decimation are well known techniques in digital signal processing as described in Discrete Time Signal Processing by Oppenheim & Schaefer. For example, assuming 2 seconds worth of an input speech is sampled at 8 kHz, where the signal has significant frequency components between 0 and 4000 Hz. Assuming the input speech signal is time-scale compressed by a factor of 2. The resultant signal would have a length of 1 second, but would still have significant frequency components between 0 and 4000 Hertz. The signal is interpolated (See Oppenheim & Schaefer) by a factor of $\alpha = 2$. This would result in a signal which is 2 seconds long, but with frequency component between 0 and 2000 Hertz. Returning to the time scale domain can be achieved by decimating the frequency compressed signal by a factor of $\alpha = 2$ to obtain the original time scaled speech (frequency components between 0-4000 Hertz) without any loss of information content.

Referring to FIGs. 18-22, timing diagrams for several iterations of the WSOLA-SD time-scaling (compression) method is shown in accordance with the present invention. Assuming that the input speech signals are appropriately digitized and stored, FIG. 18 illustrates the first iteration of the WSOLA-SD method on an uncompressed speech input signal. The WSOLA-SD method also requires the determination of an approximate pitch period of the voiced portions of the input speech signal. A brief description of the pitch determination and how the segment size is obtained from it is given below.

1) Frame input speech into 20ms blocks.
2) Compute energy in each block.
3) Compute average energy per block.
4) Determine energy threshold to detect voiced speech as a function of the average energy per block.

5) Using the energy threshold determine contiguous blocks of voiced speech of a length of at least 5 blocks.

6) On each block of the contiguous voice speech found in step 5, do a pitch analysis. This could be done using a variety of methods including Modified Auto correlation method, AMDF or Clipped auto correlation method.

7) The pitch values are smoothened using a median filter to eliminate errors in the estimation.

8) Average all the smoothened pitch values to obtain an approximate estimate of the speaker's pitch

9) Thus, the Segment size *Ss* computation is given below.

If pitch *P* greater than 60 samples $Ss = 2$*Pitch
If pitch *P* is between 40 and 60 samples $Ss = 120$
If *P* less than 40 samples $Ss = 100$
A sampling rate of 8 Khz is assumed in all cases above.

A critical factor that provides WSOLA-SD with the advantages that overcomes some of the drawbacks previously described above in the description of WSOLA is the degree of overlap f. If the degree of overlap f in WSOLA-SD is greater than 0.5, then this provides higher quality at the expense of more complexity. If the degree of overlap f in WSOLA-SD is less than 0.5, then this reduces complexity of the algorithm at the expense of quality. Thus, the user has more flexibility and control in design and use of their particular application.

Again, referring to FIGs. 18-23, the WSOLA-SD method requires a time scale factor of α (which we assume is equal to 2 for this example, where if $\alpha>1$ we have compression and if $\alpha<1$ we have expansion) and an analysis segment size (Ss) which is optimized to the input speech characteristics, namely the pitch of the speaker. An overlap segment size So is computed as f*Ss and is fixed in WSOLA-SD for a given pitch period and f. In the example shown, f is greater than 0.5, to show higher quality resultant output speech. The first Ss samples are copied directly to the output. Let the index of the last sample be I_{f1}. An overlap index O_1 is determined as So samples from the end of the last available sample in the output. Now the samples which would be overlap added are between O_1 and I_{f1}

as shown in FIG. 19. The first search index (S_1) is determined as α^*O_1 as seen in FIG. 18. After an initial portion of the input signal is copied into the output, a determination is made as to the location of the moving window of samples from the input speech signal. The window is determined around or about the search index S_1. Within the window, the best correlating So samples are determined using the cross-correlation equation previously described above, where the beginning of the best samples determined is B_1. The So samples beginning at B1 are then multiplied by an increasing ramp function (although other weighting functions could be used) and added to the last So samples in the output. Prior to the addition, the So samples in the output are multiplied by a decreasing ramp function. The resulting samples of the addition will replace the last So samples in the input. Finally, the next Ss-So samples which immediately follow the prior best matching So samples are then copied to the end of the output for use in the next iteration. This would be the end of the first iteration in WSOLA-SD.

Referring to FIGs. 20 and 21 for the next iteration, we need to compute a new overlap index O_2, similarly to O_1. Likewise, a new search index S_2 and corresponding search window is determined as done in the previous iteration. Once again, within the search window, the best correlating So samples are determined using the cross-correlation equation previously described above, where the beginning of the best samples determined is B_2. The So samples beginning at B2 are then multiplied by an increasing ramp function and added to the last So samples in the output. Prior to the addition, the So samples in the output are multiplied by a decreasing ramp function. The resulting samples of the addition will replace the last So samples in the input. Finally, the next Ss-So samples which immediately follow the prior best matching So samples are then copied to the end of the output for use in the next iteration.

FIG. 22 shows a resultant output signal from two iterations using the WSOLA-SD method. Note that there is a region of overlap (Ss-So) in the resultant output signal which insures increased intelligibility and prevents the method from skipping critical input speech components as compared to the WSOLA method.

38

Referring to FIGs. 23 and 24, an i^{th} iteration of an example input timing diagram and output timing diagram for time-scale expansion using the WSOLA-SD method is shown in accordance with the present invention. The method for expansion essentially functions similarly to the examples shown in FIGs. 18-22 except that O_i, the overlap index, moves faster than the S_i, the Search index. To be exact, O_i moves α times faster than S_i during expansion. The analysis segment size Ss is dependent on the pitch period of the input speech. The degree of overlap can range from 0 to 1, but 0.7 is used for this example in FIGs. 23 and 24. The time scaling factor α, in this instance, will be the inverse of the expansion rate. Assuming the expansion rate was 2, then the time scaling factor α = 0.5. The overlap segment size So would equal f*Ss or the degree of overlap times the analysis segment size. Thus, after several iterations of overlap adding and using an increasing ramp function on each best matching input segment and using a decreasing ramp function on each output overlap segment, prior to the addition, the input speech signal is expanded as the output speech signal that maintains all the advantages of WSOLA-SD as previously described.

Further improvement is obtained by dynamically adapting the segment size *Ss* in the WSOLA-SD algorithm with the pitch of the segment at that instant. This is done by a modification of the scheme explained previously. If we use a short segment size of *Ss* = 100 (sampling rate 8 Khz is assumed) for unvoiced speech sounds their quality is improved and for voiced speech the segment size will be *Ss* = 2*Pitch. Also a few changes are necessary to determine whether the speech segment is voiced or unvoiced. The method with these changes is described below.

1) Frame input speech into 20ms blocks.
2) Compute energy in each block.
3) Compute number of zero-crossings in each block.
4) Compute average energy per block.
5) Determine energy threshold to detect voiced speech as a function of the average energy per block.

39

5) Using the energy threshold and zero-crossing threshold determine contiguous blocks of voiced speech of length of at least 5 blocks.

6) Do pitch analysis on all the voiced segments and determine the average pitch in each of those voiced segments. This could be done using a variety of methods including Modified Auto correlation method, AMDF or Clipped auto correlation method.

7) The segments that are not marked as Voiced speech are now marked as tentative unvoiced segments.

8) Contiguous blocks of at least 5 frames in the 'tentative unvoiced segments' are taken and pitch analysis is done. The ratio of the maximum to minimum correlation coefficient is determined. If the ratio is large then the segment is classified as Unvoiced or if it is small these segments are marked as voiced and average pitch of those segments are determined along with the start and ending of the speech segment.

9) Segment size *Ss* for each of these classified speech segments are determined as follows.

If Voiced *Ss* = 2*Pitch

If Unvoiced *Ss* = 100 (Sampling rate of 8 Khz is assumed)

10) Now WSOLA-SD method of time-scaling is done, but with a varying segment size. Here the position of the input speech segment used in the processing at each time instant is determined. Depending on its position, the segment sizes *Ss* already determined is used in the processing. Using this technique results in a higher quality time-scaled speech signal.

If WSOLA-SD is used to do both compression and then a subsequent expansion on the same speech input signal as in the case of our communication system, the quality of the reconstructed speech signal can be further improved for a given average time-scale factors using several techniques.

From perceptual tests, it can be seen that a speech signal which has a higher fundamental frequency (lower pitch period) can be compressed more for a given speech quality as compared to a speech signal which has a lower fundamental frequency (higher

40

pitch period). For instance, children and female speakers will on average have a higher fundamental frequency. Thus, their speech can be compressed/expanded by 10% more without noticeably affecting the quality of their speech. Whereas male speakers who have speech on average with a lower fundamental frequency, can have their speech compressed/expanded by 10% less. Thus, in a typical communication system having roughly equal number of speakers having higher and lower fundamental frequencies, an overall improved quality in the reproduction of speech is obtained with the same compression/expansion (time-scaling) factor as before.

Another characteristic of expansion and compression using this technique leads to further enhancements. For instance, it was noticed that most of the artifacts in the speech are produced during the time-scale expansion of the speech signal. The more the speech signal is expanded the more the artifacts. It was also observed that if the speech signal is played back a little faster (less than 10%) than the original speech, the change in speed is hardly noticeable, but with a noticeable reduction in artifacts. This property helps expand the speech signal with a smaller expansion factor and thus reduce the artifacts and improve its quality. For example, if the input speech is compressed by a time-scaling factor of 3, then during expansion it would be expanded by a factor of 2.7, which means that the speech will be played faster by 10%. Since this change in speech rate will not be noticeable and reduces artifacts, it should be implemented in the method of the present invention in applications where the accuracy of the speech is not absolutely critical.

What is claimed is:

41

Claims

1. A method for compressing a plurality of voice signals within a voice communication resource having a given bandwidth within a voice communication system, comprising the steps of:

B1 5 ~~(a)~~ subchanneling the voice communication resource and simultaneously placing at least one of each of the plurality of voice signals on a subchannel~~; and~~

B2 ~~(b) compressing the time of each of the voice signals within each of the subchannels, wherein the result of steps (a) and (b) provides a compressed voice signal.~~

~~2. The method of claim 1, wherein the step of subchanneling further comprises the step of placing pairs of the plurality of voice signals on a subchannel using single sideband modulation.~~

~~3. The method of claim 1, wherein the step of subchanneling further comprises the step of modulating each of the plurality of voice signals about a plurality of pilot signals within each of the subchannels within the voice communication resource.~~

2 ~~4.~~ The method of claim 1, wherein the step of subchanneling further comprises the step of using quadrature amplitude modulation.

3 ~~5.~~ The method of claim 1, wherein the step of compressing the time of each of the voice signals further comprises the step using time-scale compression on the voice signals.

4 ~~6.~~ The method of claim 1, wherein the step of compressing the time of each of the voice signals further comprises the step of using Waveform Similarity based Overlap-Add (WSOLA) time compression on the voice signals.

42

5. ~~7.~~ The method of claim 1, wherein the step of compressing the time of each of the voice signals further comprises the speaker dependent steps of identifying pitch periods within each of the voice signals and transmitting data from one pitch period to alter a time-scaling factor.

6. ~~8.~~ The method of claim 1, wherein the step of compressing the time of each of the voice signals comprises the step of using a speaker dependent modification of the Waveform Similarity based Overlap-Add (WSOLA) time compression technique on the voice signals.

7. ~~9.~~ A method for compressing a plurality of voice signals within a voice communication resource within a voice communication system, comprising the steps of:

Ins. B3 ~~(a) compressing the bandwidth of plurality of voice signals by subchanneling the voice communication resource and modulating each of the plurality of voice signals about a plurality of pilot signals within the voice communication resource;~~

Ins. B4 ~~(b) compressing the time of the voice signal, wherein the result of steps (a) and (b) provides a compressed voice signal for transmission via a transmitter.~~

8. ~~10.~~ The method of claim 7 ~~9~~, wherein the method further comprises the step at the transmitter of transmitting the compressed voice signal to a plurality of selective call receivers.

9. ~~11.~~ The method of claim 7 ~~9~~, wherein the method further comprises the step of receiving the compressed voice signal and demodulating the compressed bandwidth signals at one of the the plurality of selective call receivers.

47

~~12~~ 10. A communication system using voice compression having at least one transmitter base station and a plurality of selective call receivers, comprising:

at the transmitter base station:

an input device for receiving an audio signal;

a processing device for compressing the audio signal using time-scale compression and a single side band modulation technique to provide a processed signal; and

Ins. B5 a transmitter for transmitting the processed signal;

at each of the plurality of selective call receivers:

a selective call receiver for receiving the transmitted processed signal;

Ins. B6 a processing device for demodulating the received processed signal using single side band demodulation and time-scale expansion to provide a reconstructed signal; and

an amplifier for amplifying the reconstructed signal into an reconstructed audio signal.

~~13~~ 11. The communication system of claim 12, wherein the single sideband modulation technique provides for the transmission of a single message split between an upper sideband and a lower sideband.

~~14~~ 12. The communication system of claim 12, wherein the single sideband modulation technique provides for the transmission of a single message repeated on an upper sideband and lower sideband.

~~15. The communication system of claim 12, wherein the system further comprises:~~

~~at the transmitter:~~

~~pilot carrier signal generator to serve as an amplitude and phase reference for distortion that occurs as a result of channel aberrations;~~

~~at the receiver;~~

~~a receiver circuit for detecting, filtering and responding to the amplitude and phase reference generated by the pilot carrier signal generator.~~

13

~~16~~. A selective call receiver for receiving compressed voice signals, comprising:

a selective call receiver for receiving a transmitted processed signal that includes compressed voice signals that have been compressed using time-scale compression;

Ins. B9 ~~a processing device for demodulating the received processed signal using single side-band demodulation and time-scale expansion to provide a reconstructed signal; and~~

Ins. B10 an amplifier for amplifying the reconstructed signal into an reconstructed audio signal.

~~17~~. The selective call receiver of claim 16, wherein the selective call receiver further comprises:

a receiver circuit for detecting, filtering and responding to the amplitude and phase reference generated by a pilot carrier signal generator in a transmitter at a base station.

14

~~18~~. A selective call paging base station for transmitting selective call signals on a communication resource having a predetermined bandwidth, comprising:

an input device for receiving a plurality of audio signals;

a means for subchannelizing the communication resource into a predetermined number of subchannels;

an amplitude compression and filtering module for each subchannel for compressing the amplitude of the respective audio signal and filtering the respective audio signal ;

a time compression module for compressing the time of the respective audio signal for each subchannel; and

a quadrature amplitude modulation transmitter for transmitting the processed signal.

15

~~19~~. The selective call paging base station of claim 18, wherein the input device for receiving a plurality of audio signals comprises a paging terminal for receiving phone messages or data messages from a computing device.

14

45

14

16 ~~20~~. The selective call paging base station of claim ~~18~~, wherein the amplitude compression and filtering module comprises an anti-alias filter coupled to an analog-to-digital converter coupled to a band-pass filter coupled to an automatic gain controller and clipper circuit.

17 ~~21~~. The selective call paging base station of claim 18, wherein the time compression module comprises a processing device for compressing the audio signal using a time-scale compression technique.

18 ~~22~~. The selective call paging base station of claim 18, wherein the time compression module comprises a processing device for compressing the audio signal using a WSOLA time compression technique.

19 ~~23~~. A selective call receiver unit for receiving compressed voice selective call signals, comprising:

a receiver having a analog to digital converter for providing a digitized received signal;

J.S. ~~a digital signal processor for performing single sideband demodulation and at least one of the functions of filtering a pilot carrier, performing automatic gain control using a feedforward loop, or decompanding of the digitized received signal to provided a processed signal; and~~

a digital to analog converter and reconstruction filter for converting the processed signal into a digitized audio signal; and

an amplifier for amplifying the digitized audio signal.

46

20 ~~24~~. A communication base station, comprising:

a terminal for receiving an audio speech signal;

an analog to digital converter for converting the audio speech signal into a digitized speech signal;

a digital signal processor for processing the digitized speech signal by performing the function of splitting the digitized speech signal and at least one of the functions of bandpass filtering, automatic gain control , time scaling, companding, or buffering; and

a transmitter having at least a Hilbert transform filter coupled to a digital to analog converter coupled to a reconstruction filter coupled to a quadrature amplitude modulator which is coupled to a radio frequency power amplifier.

08/764656

VOICE COMPRESSION METHOD AND APPARATUS IN A COMMUNICATION SYSTEM

Abstract of the Disclosure

543 680

The present invention comprises a method for compressing a plurality of voice signals within a voice communication resource (See FIG. 6) having a given bandwidth within a voice communication system. The method comprises the steps of subchanneling the voice communication resource (441, 442, 443) while placing at least one of each of the plurality of voice signals on a subchannel and time-scaling (18) each of the voice signals within each of the subchannels, wherein these steps provide a compressed voice signal.

PATENT APPLICATION DECLARATION COMBINED WITH POWER OF ATTORNEY

_X_ REGULAR (UTILITY) OR ___ DESIGN APPLICATION
(check one)

Attorney Docket No. PT00600U

As a below-named inventor, I hereby declare that:

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

I believe I am an original, first and joint inventor of the subject matter which is claimed and for which a patent is sought on the invention entitled: VOICE COMPRESSION METHOD AND APPARATUS IN A COMMUNICATION SYSTEM, the specification of which:

(check one)

_X_ is attached hereto.

___ was filed on ___________ as U.S. Application Serial No. ___________ and was amended on ___________ (if applicable).

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

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

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

Prior Foreign Application(s):

(check one)

_X_ no such applications filed.

___ such applications identified as follows:

			Priority Claimed
(Serial No.)	(Country)	(Day/Month/Year Filed)	Yes No
(Serial No.)	(Country)	(Day/Month/Year Filed)	Yes No
(Serial No.)	(Country)	(Day/Month/Year Filed)	Yes No

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

PT00600U

Send Written Correspondence To:

John H. Moore
MOTOROLA, INC.
1500 Gateway Blvd. MS 96
Boynton Beach, FL 33426-8292

Direct Telephone Calls to: (407) 739-8630

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 statement 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.

Full name of first joint inventor Clifford Dana Leitch 1-00

Inventor's signature Clifford Dana Leitch Date 2-27-95

Residence Coral Springs, FL Florida
City State or Foreign Country

Citizenship USA

Post Office Address 673 NW 11th Way
Street Address

Coral Springs, FL 33071
City State or Country Zip Code

Full name of second inventor Robert John Schwendeman 2-00

Inventor's signature Robert John Schwendeman Date 2/27/95

Residence Pompano Beach, FL Florida
City State or Foreign Country

Citizenship USA

Post Office Address 590 S.E. 10th Avenue
Street Address

Pompano Beach, FL 33060
City State or Country Zip Code

PT00600U

Full name of third inventor Kazimierz Siwiak 3-00

Inventor's signature Kazimierz Siwiak Date 27 Feb 95

Residence Coral Springs, FL Florida
City / State or Foreign Country

Citizenship USA

Post Office Address 10988 N.W. 14th Street
Street Address

Coral Springs, FL 33071
City / State or Country / Zip Code

Full name of fourth inventor William Joseph Kuznicki 4-00

Inventor's signature William Joseph Kuznicki Date 2/27/95

Residence Coral Springs, FL Florida
City / State or Foreign Country

Citizenship USA

Post Office Address 3681 N.W. 100th Avenue
Street Address

Coral Springs, FL 33065
City / State or Country / Zip Code

Full name of fifth inventor Sunil Satyamurti 5-00

Inventor's signature S. Sunil Date 02/28/95

Residence Delray Beach, FL FL
City / State or Foreign Country

Citizenship India

Post Office Address 1935-D Linton Lake Drive
Street Address

Delray Beach, Florida 33445
City / State or Country / Zip Code

08/395747

#2
Prior Art
cw 6/11/95

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re application of: : Date: February 28, 1995

LEITCH, et al. :

Docket No.: PT00600U :

Filed: Concurrently Herewith :

For: VOICE COMPRESSION METHOD AND APPARATUS IN A COMMUNICATION SYSTEM

INFORMATION DISCLOSURE STATEMENT
PURSUANT TO 37 C.F.R. §§ 1.56, 1.97, and 1.98

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

Sir:

Applicant submits herewith the art listed below of which the Applicant is aware, which the Applicant believes may be material to the examination of the subject application and in respect of which there may be a duty to disclose in accordance with 37 C.F.R. § 1.56. This citation of information, also appearing on the attached Form PTO-1449, "List of Art Cited by Applicant" is made pursuant to 37 C.F.R. §§ 1.56, 1.97, and 1.98.

A copy of the art listed below is enclosed herewith, unless the art is cumulative as noted below or the art has been cited in a prior application from which an earlier filing date is claimed for the subject application, the earlier application noted below.

The filing of this Information Disclosure Statement shall not be construed as a representation that a search has been made, an admission that the information cited is, or is considered to be, material to patentability, or that no other material information exists. Further, the filing of this Information Disclosure Statement shall not be construed as an admission against interest in any manner.

Pursuant to 37 C.F.R. § 1.98, as amended March 16, 1992, no explanation of the relevance of the English language references is presented.

Patents

U.S. Patent No. 5,239,306 issued August 24, 1993 to Siwiak, et al.

U.S. Patent No. 4,839,923 issued June 13, 1989 to Kotzin.

U.S. Patent No. 5,068,898 issued November 26, 1991 to Dejmek, et al.

U.S. Patent No. 5,175,769 issued December 29, 1992 to Hejna, Jr., et al.

U.S. Patent No. 5,216,744 issued June 1, 1993 to Alleyne, et al.

U.S. Patent No. 5,282,205 issued January 25, 1994 to Kuznicki.

Publications

Verhelst and Roelands, An Overlap-Add Technique Based On Waveform Similarity (WSOLA) For High Quality Time-Scale Modification Of Speech, IEEE 1993, p. II-554 - II-557.

Oppenheim and Schafer, Changing the Sampling Rate Using Discrete-Time Processing, Discrete-Time Signal Processing, 1989, Ch. 3.6, p. 101-112.

The attorney signing below is making this Information Disclosure Statement on the basis of information supplied by the inventor, an individual associated with the filing and prosecution of the subject application and/or information in the attorney's files. The citation of this information does not constitute either an admission of priority or a waiver of any right applicant may have under applicable statutes, Rules of Practice in patent cases, or otherwise.

Respectfully submitted,

LEITCH, et al.

By: Pablo Meles

Pablo Meles
Attorney for Applicant
Registration No. 33,739
Tel. (407) 739-2860
Fax. (407) 739-2825

MOTOROLA, INC.
Patent Department, MS96
1500 Gateway Boulevard
Boynton Beach, FL 33426-8292

#2

FORM PTO-1449 U.S. DEPARTMENT OF COMMERCE, PATENT AND TRADEMARK OFFICE

LIST OF ART CITED BY APPLICANT

(Use several sheets if necessary)

ATTY. DOCKET NO.	SERIAL NO.	FILING DATE
PT00600U	08/764856	Concurrently herewith
APPLICANT		GROUP
LEITCH, et al.		

U.S. PATENT DOCUMENTS

EXAMINER INITIAL		DOCUMENT NUMBER	DATE	NAME	CLASS	SUBCLASS	FILING DATE IF APPROPRIATE
E	AA	5,239,306	8/24/93	Siwiak, et al.	340	825.44	
E	AB	4,839,923	6/13/89	Kotzin	381	31	
E	AC	5,068,898	11/26/91	Dejmek, et al.	381	29	
E	AD	5,175,769	12/29/92	Hejna, Jr., et al.	340	825.44	
E	AE	5,216,744	6/01/93	Alleyne, et al.	381	29	
E	AF	5,282,205	1/25/94	Kuznicki	370	94.1	
	AG						
	AH						
	AI						
	AJ						
	AK						

FOREIGN PATENT DOCUMENTS

		DOCUMENT NUMBER	DATE	COUNTRY	CLASS	SUBCLASS	TRANSLATION YES	TRANSLATION NO.
	AL							
	AM							
	AN							
	AO							

OTHER PRIOR ART *(Including Author, Title, Date, Pertinent Pages, Etc.)*

E	AP	Verhelst and Roelands, An Overlap-Add Technique Based On Waveform Similarity (WSOLA) For High Quality Time-Scale Modification Of Speech, IEEE 1993, p. II-554 - II-557.
	AR	Oppenheim and Schafer, Changing the Sampling Rate Using Discrete-Time Processing, Discrete-Time Signal Processing, 1989, Ch. 3.6, p. 101-112.
	AS	
	AT	

EXAMINER [signature] DATE CONSIDERED 3/15/96

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

AN OVERLAP-ADD TECHNIQUE BASED ON WAVEFORM SIMILARITY (WSOLA) FOR HIGH QUALITY TIME-SCALE MODIFICATION OF SPEECH

Werner VERHELST and Marc ROELANDS

Vrije Universiteit Brussel
Faculty of Applied Science, dept. ETRO/DSSP
Pleinlaan 2, B-1050 Brussels
Belgium

ABSTRACT

A concept of waveform similarity is proposed for tackling the problem of time-scale modification of speech, and is worked-out in the context of short-time Fourier transform representations.

The resulting WSOLA algorithm produces high quality speech output, is algorithmically and computationally efficient and robust, and allows for on-line processing with arbitrary time-scaling factors that may be specified in a time-varying fashion and that can be chosen over a wide continuous range of values.

I. INTRODUCTION

Algorithms for high quality time-scale modification of speech are important for applications such as voice-mail and dictation-tape playback or post synchronization of film and video, where the potentiality of controlling the apparent speaking rate is a desirable feature. The problem with time-scaling a speech signal $x(n)$ lies in realising the specified time-warp function $\tau(n)$ in such a way as to affect the apparent speaking rate only, preserving other perceived aspects such as timbre, voice quality, and pitch. We will therefore consider in this paper that the ideal time-scaling algorithm should produce a synthetic waveform $y(n)$ that maintains maximal local similarity to the original waveform $x(m)$ in corresponding neighbourhoods of related sample indices $n = \tau(m)$. This could be expressed mathematically as

$$\forall m: y(n+\tau(m)).w(n) \simeq x(n+m).w(n), \qquad (1)$$

where $w(n)$ is a windowing function, and the symbol '$\simeq$' is defined to hold the rather vague meaning 'maximally similar to'.

Assuming that the relation of maximal similarity persists after Fourier transformation, and defining the short-time Fourier transform $X(\omega,m)$ of a sequence $x(m)$ by

$$X(\omega,m) = \sum_{n=-\infty}^{\infty} x(n+m).w(n).e^{-j\omega n},$$

expression (1) can be rewritten as

$$Y(\omega,\tau(m)) \simeq X(\omega,m). \qquad (2)$$

If we choose the effective length of $w(n)$ in (1) to span at least one pitch period, we can expect that the important perceptual characteristics of the signal can remain fairly unaffected by the time-scaling operation, provided that $Y(\omega,m)$ can be specified in accordance with a suitable similarity measure.

In general, finding an operational definition for '$\simeq$' in (2) amounts to solving the time-scaling problem based on manipulation of short-time Fourier transforms. Section 2 of this paper discusses some of the problems encountered and the approach taken in algorithms of the overlap-add (OLA) and synchronized overlap-add (SOLA) traditions. Section 3 introduces our WSOLA approach as a variant in which we explicitly pursued the idea of making operational the intuitive notion of maximal local waveform similarity. Before concluding the paper, we indicate in section 4 that WSOLA produces a natural sounding output and is algorithmically and computationally efficient and robust, and allows for on-line processing with arbitrary time-scaling factors that may be specified in a time-varying fashion and can be chosen over a wide continuous range of values.

II. TIME-SCALE MODIFICATION BASED ON OLA-TECHNIQUES

Let $X(\omega,\tau^{-1}(L_k))$ represent a down-sampled version of the short-time Fourier transform (STFT) of the input signal $x(n)$, and assume we force $\simeq$ to represent strict equality in equation (2) by specifying the 2-dimensional function

$$\hat{Y}(\omega,L_k) = X(\omega,\tau^{-1}(L_k)). \qquad (3)$$

It will be clear that, except for some trivial cases such as $w(n) = 0$ or $\tau(m) = m$, there will not generally exist a solution for equation (1) with equality required. This implies that there will not generally exist a signal $\hat{y}(n)$ that has $\hat{Y}(\omega,L_k)$ as a STFT or, equivalently, that $\hat{Y}(\omega,L_k)$ is not valid as a STFT.

This kind of problem is liable to occur whenever the intent is to create a 1-dimensional signal by constructing a 2-dimensional STFT-representation of it. As a possible solution, the overlap-add technique [1] proposes to synthesize a signal $y(n)$ whose STFT $Y(\omega,L_k)$ is as close as possible to the desired $\hat{Y}(\omega,L_k)$ in the least-squares (LS) sense.

0-7803-0946-4/93 $3.00

In case $\hat{Y}(\omega, L_k)$ is a time-warped STFT, as in (3), the corresponding synthesis equation becomes

$$y(n) = \frac{\sum_k v(n-L_k).x(n+\tau^{-1}(L_k)-L_k)}{\sum_k v(n-L_k)}, \qquad (4)$$

where $v(n) = w^2(n)$ is a windowing function and the L_k represent consecutive window positions. The basic OLA-synthesis operation (numerator of eq. 4) then consists of cutting-out input segments around analysis instants $\tau^{-1}(L_k)$, and repositioning them at corresponding synthesis instants L_k before adding them together to form the output signal. While it can be noted that straightforward application of the OLA procedure to the time-warped STFT $\hat{Y}(\omega, L_k)$ corresponds to interpreting $\Leftarrow$ as 'maximally close in LS-sense' in $Y(\omega, L_k) \Leftarrow X(\omega, \tau^{-1}(L_k))$ (a down-sampled version of eq. (2)), it would not be an appropriate choice in this case. It was shown in [2] that this is not due to the OLA procedure itself, but rather to the choice that was made for $\hat{Y}(\omega, L_k)$. Indeed, it was expressed in eq. (3) that individual output segments should ideally correspond to input segments that have been repositioned according to the desired time-warp function. As a result, the OLA procedure in eq. (4) does reposition the individual input segments with respect to each other (destroying original phase relationships in the process) and constructs the output signal by interpolating between these misaligned segments. The resulting distortions are detrimental for signal quality and are illustrated in figure 1.

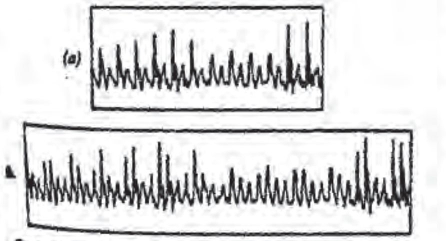


Fig. 1. *OLA-synthesis from the time-warped STFT does not succeed to replicate the quasi-periodic structure of the original signal (a) in its output (b).*

To avoid pitch period discontinuities or phase jumps at waveform-segment joins, [2] proposes to realign each input segment to the already formed portion of the output signal before performing the OLA operation. The resulting synchronized OLA algorithm (SOLA) thus produces the signal

$$y(n) = \frac{\sum_k v(n-L_k+\Delta_k).x(n+\tau^{-1}(L_k)-L_k+\Delta_k)}{\sum_k v(n-L_k+\Delta_k)}, \qquad (5)$$

in a left-to-right fashion, where shift factors Δ_k ($\in[-\Delta_{max}..\Delta_{max}]$) are chosen such as to maximize the cross-correlation coefficient between $v(n-L_k+\Delta_k).x(n+\tau^{-1}(L_k)-L_k+\Delta_k)$ and

$$y_{k-1}(n) = \frac{\sum_{l=-\infty}^{k-1} v(n-L_l+\Delta_l).x(n+\tau^{-1}(L_l)-L_l+\Delta_l)}{\sum_{l=-\infty}^{k-1} v(n-L_l+\Delta_l)}.$$

Another form of synchronization is obtained by applying time-domain pitch-synchronized OLA technique (TD-PSOLA [3]). In that case the OLA procedure is performed pitch synchronously (i.e., L_k–L_{k-1} equals the local pitch period) on segments that are, accordingly, excised in a pitch synchronous way from an original $x(n)$.

We can thus observe that both SOLA and TD-PSOLA recognize that a tolerance Δ_k is needed in order to ensure proper segment synchronization in OLA synthesis: while SOLA uses this tolerance to allow for post-synchronization in

$$\hat{Y}(\omega, L_k+\Delta_k) = X(\omega, \tau^{-1}(L_k)),$$

TD-PSOLA uses it in a pre-synchronization step to obtain a pitch synchronous STFT on both sides of

$$\hat{Y}(\omega, L_k) = X(\omega, \tau^{-1}(L_k)+\Delta_k).$$

III. WSOLA: AN OVERLAP-ADD TECHNIQUE BASED ON WAVEFORM SIMILARITY

It was shown in the preceding section that, if an OLA synthesis procedure is to be used for time-scaling, one should allow for a tolerance on the time-warping function that will actually be realised. In fact, this tolerance can be seen to give concrete form to the words 'corresponding neighbourhoods' that were used in the introductory section to state that 'the ideal time-scaling algorithm should produce a synthetic waveform $y(n)$ that maintains maximal local similarity to the original waveform $x(n)$ in corresponding neighbourhoods of related sample indices $n = \tau(m)$'.[1]

Like TD-PSOLA, WSOLA uses this timing tolerance for specifying the input segments that are to be used in the OLA procedure. Therefore, in both cases the basic synthesis equation is

$$y(n) = \frac{\sum_k v(n-L_k).x(n+\tau^{-1}(L_k)+\Delta_k-L_k)}{\sum_k v(n-L_k)}, \qquad (6)$$

While in TD-PSOLA the Δ_k are chosen such that pitch synchronicity is maintained, WSOLA uses them to ensure that the time-scale modified waveform can maintain maximal similarity to the original (natural) waveform across its segment joins. In other words, WSOLA ensures sufficient signal continuity at segment joins by requiring maximal similarity to the natural continuity that existed in the input signal. Based on

[1] A better mathematical rendering of the intended meaning would have been possible by using

$$\forall m: y(n+m).w(n) \Leftarrow x(n+\tau^{-1}(m)+\Delta_m).w(n)$$

instead of eq. (1).

this idea, a variety of practical implementations can be constructed. The operation of a basic version of the WSOLA technique is illustrated in figure 2 and explained below.

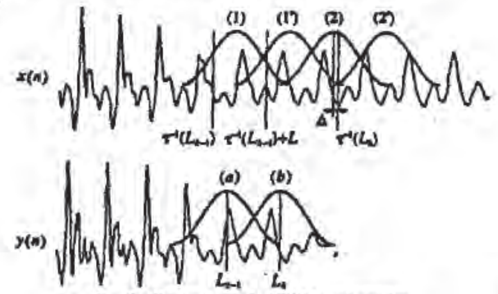


Fig. 2. *Illustration of a WSOLA algorithm.*

By choosing regularly spaced synthesis instants $L_k = kL$ and a symmetric window such that [2]

$$\sum_k v(n-kL)=1, \tag{7}$$

synthesis equation (6) simplifies to

$$y(n)=\sum_k v(n-kL).x(n+\tau^{-1}(kL)-kL+\Delta_k). \tag{8}$$

Proceeding in a left-to-right fashion, assume segment (1) from figure 2 was the last segment that was excised from the input and added to the output at time instant $L_{k-1} = (k-1)L$, i.e. segment (a) = segment (1). WSOLA then needs to find a segment (b) that will overlap-add with (a) in a synchronized way and can be excised from the input around time instant $\tau^{-1}(kL)$. As (1') would overlap-add with (1) = (a) in a natural way to form a portion of the original input speech, WSOLA can select (b) such that it resembles (1') as closely as possible and is located within the prescribed tolerance interval around $\tau^{-1}(kL)$ in the input wave. The position of this best segment (2) is found by maximizing a similarity measure (such as the cross-correlation or the cross-AMDF) between the sample sequence underlying (1') and the input speech. After overlap-adding (b) with (a), WSOLA proceeds to the next output segment, where (2') now plays the same role as (1') in the previous step.

Figure 3 illustrates in more detail how the position of a best segment m is determined by finding the value $\delta = \Delta_m$ that lies within a tolerance region $[-\Delta_{max}..\Delta_{max}]$ around $\tau^{-1}(mL)$ and maximizes the chosen similarity measure $c(m,\delta)$ with respect to the signal portion that would form a natural continuation for the previously chosen segment m-1.

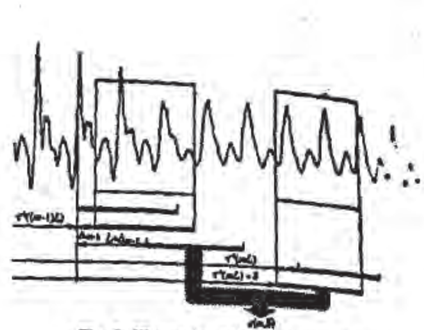


Fig. 3. *Illustration of similarity-based signal segmentation in WSOLA.*

N representing the window length, some examples of similarity measures that can be applied successfully are:

- a cross-correlation coefficient

$$c_c(m,\delta)=\sum_{n=0}^{N-1} x(n+\tau^{-1}((m-1)L)+\Delta_{m-1}+L).x(n+\tau^{-1}(mL)+\delta),$$

- a normalised cross-correlation coefficient

$$c_n(m,\delta)=\frac{c_c(m,\delta)}{\left(\sum_{n=0}^{N-1} x^2(n+\tau^{-1}(mL)+\delta)\right)^{1/2}},$$

- or a cross-AMDF coefficient

$$c_A(m,\delta)=\sum_{n=0}^{N-1}\left|x(n+\tau^{-1}((m-1)L)+\Delta_{m-1}+L)-x(n+\tau^{-1}(mL)+\delta\right|.$$

IV. EVALUATION

The performance of WSOLA was evaluated in extensive informal listening tests (many of them concerned WSOLA with 20 ms hanning windowing, 50% overlap, $\Delta_{max} = 5$ ms, $c_c(m,\delta)$ or $c_A(m,\delta)$, and 10 kHz sampling frequency). For all time-scaling factors tested ($\tau(t) = \alpha t$, with $\alpha \in [0.4 .. 0.7] \cup [1.3 .. 2.0]$) we found the resulting speech quality to be very high and to be robust against background noises, including competing voices. (Figure 4 shows an example output waveform.)

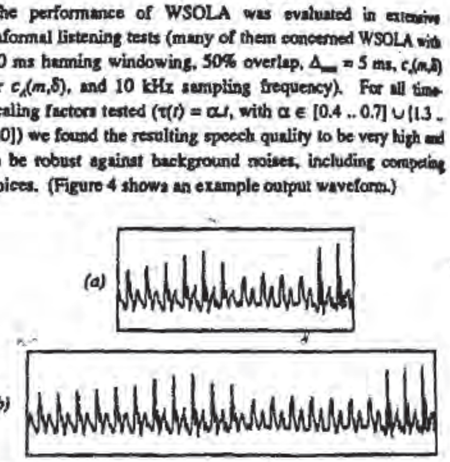


Fig. 4. *(a) original speech fragment, (b) corresponding WSOLA output waveform when slowed down with $\alpha = 0.6$.*

[2] A 30 ms hanning window with 50% overlap, for example, satisfies this condition and is a fairly standard choice.

As mentioned before, many variants of the basic WSOLA technique are possible. These can be constructed for example by varying the windowing function, the similarity measure, or the portion of $x(n)$ that is to serve as the reference for natural signal continuity. This design flexibility can be used to optimize the algorithm for implementation on a given target system. We found that all tested variants of the algorithm provided similar high quality, from which we concluded that waveform-similarity is a real powerful principle for time-scaling. As an example, figure 5 illustrates the robustness of WSOLA against the choice of distance measure.

Fig. 5. *Histogram of the difference between alignment parameters Δ_k obtained from $c_x(m,\delta)$ and from $c_A(m,\delta)$.*

While SOLA and TD-PSOLA will produce an equally high speech quality when operated properly, they each present some disadvantages compared to WSOLA. Table 1 summarizes a qualitative comparison between the three methods.

	TD-PSOLA	SOLA	WSOLA
Synchronizing method	pitch epochs	output similarity	input similarity
Effective window length	pitch adaptive	fixed (> 4.pitch)	fixed
Normalizing denominator = constant	no	no	yes
Algorithmic & computational efficiency	low	high	very high
Robustness	low	high	high
Speech quality	high	high	high
Pitch modification	yes	no	no

Table 1. *Comparison of synchronized overlap-add techniques for on-line time-scale modification of speech.*

TD-PSOLA requires means for pitch-synchronization (which is difficult to automate in a reliable way) and uses a pitch adaptive window length. In order to obtain robust synchronization with SOLA, we found that relatively long windows of about 80ms seem to be required. As WSOLA uses a synchronous segmentation technique with a fixed length window in combination with regularly spaced synthesis intervals (allowing the normalising denominator of the synthesis equation to be made constant), it is computationally and algorithmically more efficient than either SOLA or TD-PSOLA. It can be noted that the variant of TD-PSOLA that was considered can be operated in much the same way as a pitch excited vocoder [4]. While it could consequently be used for a more general prosodic modification of speech, it remains true that WSOLA can be preferred when only time-scale modification needs to be performed.

CONCLUSION

A concept of waveform similarity was proposed for tackling the problem of time-scale modification of speech, and was worked-out in the context of STFT manipulation.

The resulting WSOLA algorithm is designed in the tradition of the OLA, SOLA and TD-PSOLA techniques, and provides high quality output speech with high algorithmic and computational efficiency and robustness.

REFERENCES

[1] D.W. Griffin, J.S. Lim, 'Signal Estimation from Modified Short-Time Fourier Transforms', IEEE Trans. on Acoust., Speech, and Signal Processing, Vol. ASSP-32, No. 2, pp. 236-243, 1984.

[2] S. Roucos, A. Wilgus, 'High quality Time-Scale Modification of Speech', ICASSP-85, pp. 236-239, 1985.

[3] E. Moulines, F. Charpentier, 'Pitch-Synchronous Waveform Processing Techniques for Text-to-Speech Synthesis Using Diphones', Speech Communication Vol. 9 (5/6), pp. 453-467, 1990.

[4] W. Verhelst, 'On the Quality of Speech Produced by Impulse Driven Linear Systems', ICASSP-91, pp. 501-504, 1991.

PRENTICE HALL SIGNAL PROCESSING SERIES

Alan V. Oppenheim, Editor

ANDREWS AND HUNT *Digital Image Restoration*
BRIGHAM *The Fast Fourier Transform*
BRIGHAM *The Fast Fourier Transform and Its Applications*
BURDIC *Underwater Acoustic System Analysis*
CASTLEMAN *Digital Image Processing*
COWAN AND GRANT *Adaptive Filters*
CROCHIERE AND RABINER *Multirate Digital Signal Processing*
DUDGEON AND MERSEREAU *Multidimensional Digital Signal Processing*
HAMMING *Digital Filters, 3/E*
HAYKIN, ED. *Array Signal Processing*
JAYANT AND NOLL *Digital Coding of Waveforms*
KAY *Modern Spectral Estimation*
KINO *Acoustic Waves: Devices, Imaging, and Analog Signal Processing*
LEA, ED. *Trends in Speech Recognition*
LIM *Two-Dimensional Signal and Image Processing*
LIM, ED. *Speech Enhancement*
LIM AND OPPENHEIM, EDS. *Advanced Topics in Signal Processing*
MARPLE *Digital Spectral Analysis with Applications*
MCCLELLAN AND RADER *Number Theory in Digital Signal Processing*
M[illegible] *Lessons in Digital Estimation Theory*
O[illegible]EIM, ED. *Applications of Digital Signal Processing*
OPPENHEIM, WILLSKY, WITH YOUNG *Signals and Systems*
OPPENHEIM AND SCHAFER *Digital Signal Processing*
OPPENHEIM AND SCHAFER *Discrete-Time Signal Processing*
QUACKENBUSH ET AL. *Objective Measures of Speech Quality*
RABINER AND GOLD *Theory and Applications of Digital Signal Processing*
RABINER AND SCHAFER *Digital Processing of Speech Signals*
ROBINSON AND TREITEL *Geophysical Signal Analysis*
STEARNS AND DAVID *Signal Processing Algorithms*
TRIBOLET *Seismic Applications of Homomorphic Signal Processing*
WIDROW AND STEARNS *Adaptive Signal Processing*

Discrete-Time Signal Processing

Alan V. Oppenheim
Ronald W. Schafer

PRENTICE HALL, Englewood Cliffs, New Jersey 07632

Library of Congress Cataloging-in-Publication Data

Oppenheim, Alan V.
Discrete-time signal processing / Alan V. Oppenheim, Ronald W. Schafer.
p. cm.—(Prentice Hall signal processing series)
Bibliography: p.
Includes index.
ISBN 0-13-216292-X
1. Signal processing—Mathematics. 2. Discrete-time systems.
I. Schafer, Ronald W. II. Title. III. Series.
TK5102.5.02452 1989
621.38'043—dc 19 88-25562
CIP

Editorial/production supervision: Barbara G. Flanagan
Interior design: Roger Brower
Cover design: Vivian Berman
Manufacturing buyer: Mary Noonan

Printed in the United States of America
10 9 8 7 6 5 4 3 2

ISBN 0-13-216292-X

Prentice-Hall International (UK) Limited, *London*
Prentice-Hall of Australia Pty. Limited, *Sydney*
Prentice-Hall Canada Inc., *Toronto*
Prentice-Hall Hispanoamericana, S.A., *Mexico*
Prentice-Hall of India Private Limited, *New Delhi*
Prentice-Hall of Japan, Inc., *Tokyo*
Simon & Schuster Asia Pte. Ltd., *Singapore*
Editora Prentice-Hall do Brasil, Ltda., *Rio de Janeiro*

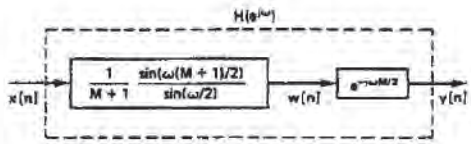


Figure 3.16 The moving average system represented as a cascade of two systems.

The noninteger delay represented by Eq. (3.64) has considerable practical significance since such a factor often arises in the frequency-domain representation of systems. When such a term is found in the frequency response of a causal discrete-time system, it can be interpreted in the light of this example. This interpretation is illustrated in Example 3.4.

Example 3.4 Moving Average

In Example 2.15 we considered the moving average system and obtained its frequency response. For the case $M_1 = 0$ and $M_2 = M$, the frequency response is

$$H(e^{j\omega}) = \frac{1}{(M+1)}\frac{\sin[\omega(M+1)/2]}{\sin(\omega/2)}e^{-j\omega M/2}, \qquad |\omega| < \pi. \tag{3.65}$$

Let us interpret this as the cascade of two systems, as indicated in Fig. 3.16. The first system imposes a frequency-domain amplitude weighting. The second system represents the linear phase term in Eq. (3.65). If M is an even integer (meaning the moving average of an odd number of samples), then the linear phase term corresponds to an integer delay, i.e.,

$$y[n] = w[n - M/2]. \tag{3.66}$$

However, if M is odd, the linear phase term corresponds to a noninteger delay, specifically, an integer plus one-half sample interval. This noninteger delay can be interpreted in terms of the discussion in Example 3.3; i.e., $y[n]$ is equivalent to bandlimited interpolation of $w[n]$ followed by a continuous-time delay of $MT/2$ (where T is the assumed but arbitrary sampling period associated with the D/C interpolation of $w[n]$) followed by C/D conversion again with sampling period T.

3.6 CHANGING THE SAMPLING RATE USING DISCRETE-TIME PROCESSING

We have seen that a continuous-time signal $x_c(t)$ can be represented by a discrete-time signal consisting of a sequence of samples

$$x[n] = x_c(nT). \tag{3.67}$$

Alternatively, our previous discussion has shown that even if $x[n]$ was not obtained originally by sampling, we can always use Eq. (3.24) to find a continuous-time bandlimited signal $x_r(t)$ whose samples are $x[n] = x_r(nT)$.

It is often necessary to change the sampling rate of a discrete-time signal, i.e., to obtain a new discrete-time representation of the underlying continuous-time signal of the form

$$x'[n] = x_c(nT'), \tag{3.68}$$

where $T' \neq T$. One approach to obtaining the sequences $x'[n]$ from $x[n]$ is to reconstruct $x_c(t)$ from $x[n]$ using Eq. (3.24) and then resample $x_c(t)$ with period T' to obtain $x'[n]$. Often this is not a desirable approach because of the nonideal analog reconstruction filter, D/A converter, and A/D converter that would be used in a practical implementation. Thus, it is of interest to consider methods of changing the sampling rate that involve only discrete-time operations.

3.6.1 Sampling Rate Reduction by an Integer Factor

The sampling rate of a sequence can be reduced by "sampling" it, i.e., by defining a new sequence

$$x_d[n] = x[nM] = x_c(nMT). \tag{3.69}$$

Equation (3.69) defines the system depicted in Fig. 3.17, which is called a *sampling rate compressor* (Crochiere and Rabiner, 1983) or simply a *compressor*. From Eq. (3.69) it is clear that $x_d[n]$ could be obtained directly from $x_c(t)$ by sampling with period $T' = MT$. Furthermore, if $X_c(j\Omega) = 0$ for $|\Omega| > \Omega_N$, then $x_d[n]$ is an exact representation of $x_c(t)$ if $\pi/T' = \pi/(MT) > \Omega_N$. That is, the sampling rate can be reduced by a factor of M without aliasing if the original sampling rate was at least M times the Nyquist rate or if the bandwidth of the sequence is first reduced by a factor of M by discrete-time filtering. In general, the operation of reducing the sampling rate (including any prefiltering) will be called *downsampling*.

As in the case of sampling a continuous-time signal, it is useful to obtain a frequency-domain relation between the input and output of the compressor. This time, however, it will be a relationship between discrete-time Fourier transforms. Although several methods can be used to derive the desired result, we will base our derivation on the results already obtained for sampling continuous-time signals. First recall that the discrete-time Fourier transform of $x[n] = x_c(nT)$ is

$$X(e^{j\omega}) = \frac{1}{T}\sum_{k=-\infty}^{\infty} X_c\left(j\frac{\omega}{T} - j\frac{2\pi k}{T}\right). \tag{3.70}$$

Similarly, the discrete-time Fourier transform of $x_d[n] = x[nM] = x_c(nT')$ with $T' = MT$ is

$$X_d(e^{j\omega}) = \frac{1}{T'}\sum_{r=-\infty}^{\infty} X_c\left(j\frac{\omega}{T'} - j\frac{2\pi r}{T'}\right). \tag{3.71}$$

Now, since $T' = MT$, we can write Eq. (3.71) as

$$X_d(e^{j\omega}) = \frac{1}{MT}\sum_{r=-\infty}^{\infty} X_c\left(j\frac{\omega}{MT} - j\frac{2\pi r}{MT}\right). \tag{3.72}$$

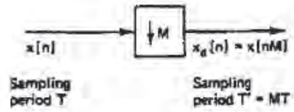


Figure 3.17 Representation of downsampler or discrete-time sampler.

To see the relationship between Eqs. (3.72) and (3.70), note that the summation index r in Eq. (3.72) can be expressed as

$$r = i + kM, \tag{3.73}$$

where k and i are integers such that $-\infty < k < \infty$ and $0 \le i \le M-1$. Clearly r is still an integer ranging from $-\infty$ to ∞, but now Eq. (3.72) can be expressed as

$$X_d(e^{j\omega}) = \frac{1}{M}\sum_{i=0}^{M-1}\left[\frac{1}{T}\sum_{k=-\infty}^{\infty} X_c\left(j\frac{\omega}{MT} - j\frac{2\pi k}{T} - j\frac{2\pi i}{MT}\right)\right]. \tag{3.74}$$

The term inside the square brackets in Eq. (3.74) is recognized from Eq. (3.70) as

$$X(e^{j(\omega-2\pi i)/M}) = \frac{1}{T}\sum_{k=-\infty}^{\infty} X_c\left(j\frac{\omega-2\pi i}{MT} - j\frac{2\pi k}{T}\right). \tag{3.75}$$

Thus we can express Eq. (3.74) as

$$X_d(e^{j\omega}) = \frac{1}{M}\sum_{i=0}^{M-1} X(e^{j(\omega/M - 2\pi i/M)}). \tag{3.76}$$

The analogy between Eqs. (3.70) and (3.76) is clear. Equation (3.70) expresses the Fourier transform of the sequence of samples $x[n]$ (period T) in terms of the Fourier transform of the continuous-time signal $x_c(t)$. Equation (3.76) expresses the Fourier transform of the discrete-time sampled sequence $x_d[n]$ (sampling period M) in terms of the Fourier transform of the sequence $x[n]$. If we compare Eqs. (3.71) and (3.76) we see that $X_d(e^{j\omega})$ can be thought of as being composed of either an infinite set of copies of $X_c(j\Omega)$, frequency scaled through $\omega = \Omega T'$ and shifted by integer multiples of $2\pi/T'$ (Eq. 3.71), or M copies of the periodic Fourier transform $X(e^{j\omega})$, frequency scaled by M and shifted by integer multiples of $2\pi/M$ (Eq. 3.76). Either interpretation makes it clear that $X_d(e^{j\omega})$ is periodic with period 2π (as are all discrete-time Fourier transforms) and that aliasing can be avoided by ensuring that $X(e^{j\omega})$ is bandlimited, i.e.,

$$X(e^{j\omega}) = 0, \quad \omega_N \le |\omega| \le \pi, \tag{3.77}$$

and $2\pi/M \ge 2\omega_N$.

Downsampling is illustrated in Fig. 3.18. Figure 3.18(a) shows the Fourier transform of a bandlimited continuous-time signal, and Fig. 3.18(b) shows the Fourier transform of the impulse train of samples obtained with sampling period T. Figure 3.18(c) shows $X(e^{j\omega})$ and is related to Fig. 3.18(b) through Eq. (3.18). As we have already seen, Figs. 3.18(b) and (c) differ only in a relabeling of the frequency axis. Figure 3.18(d) shows the discrete-time Fourier transform of the downsampled sequence when $M = 2$. We have plotted this Fourier transform as a function of normalized frequency $\omega = \Omega T'$. Finally, Fig. 3.18(e) shows the discrete-time Fourier transform of the downsampled sequence plotted as a function of the continuous-time

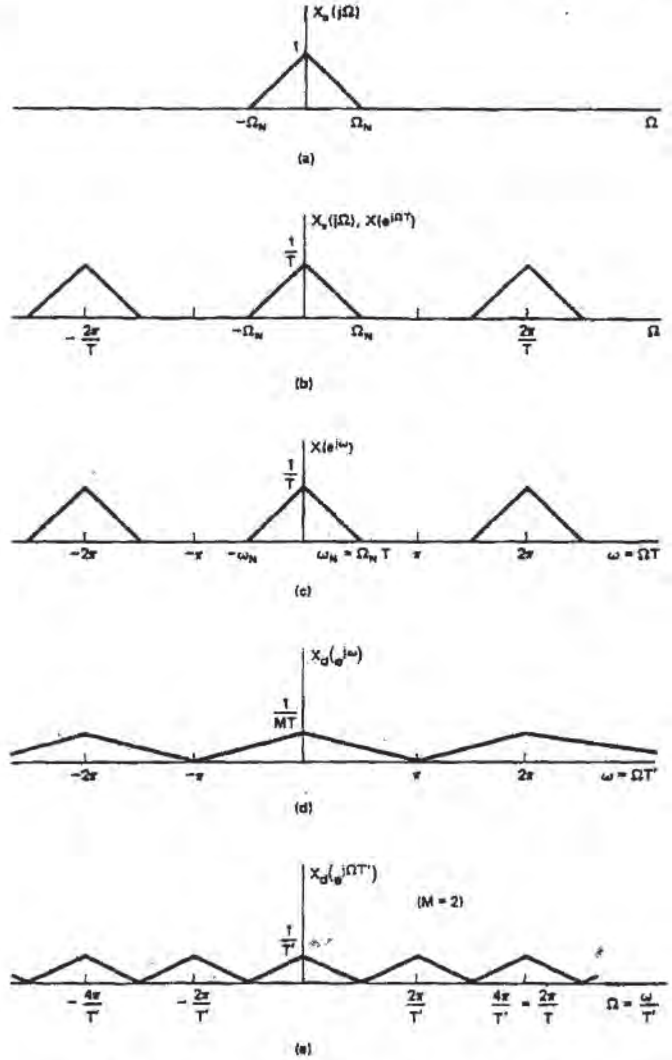


Figure 3.18 Frequency-domain illustration of downsampling.

frequency variable Ω. Figure 3.18(e) is identical to Fig. 3.18(d) except for the relabeling of the frequency axis through the relation $\Omega = \omega/T'$.

In this example, $2\pi/T = 4\Omega_N$, i.e., the original sampling rate is exactly twice the minimum rate to avoid aliasing. Thus, when the original sampled sequence is downsampled by a factor of $M = 2$, no aliasing results. If the downsampling factor is more than 2 in this case, aliasing will result, as illustrated in Fig. 3.19.

Figure 3.19(a) shows the continuous-time Fourier transform of $x_c(t)$, and Fig. 3.19(b) shows the discrete-time Fourier transform of the sequence $x[n] = x_c(nT)$, when $2\pi/T = 4\Omega_N$. Thus, $\omega_N = \Omega_N T = \pi/2$. Now if we downsample by a factor of $M = 3$, we obtain the sequence $x_d[n] = x[3n] = x_c(n3T)$ whose discrete-time Fourier transform is plotted in Fig. 3.19(c) with normalized frequency $\omega = \Omega T'$. Note that because $M\omega_N = 3\pi/2$, which is greater than π, aliasing occurs. In general, to avoid aliasing in downsampling by a factor of M requires that

$$\omega_N M < \pi \qquad \text{or} \qquad \omega_N < \pi/M. \tag{3.78}$$

If this condition does not hold, aliasing occurs, but it may be tolerable for some applications. In other cases, downsampling can be done without aliasing if we are willing to reduce the bandwidth of the signal $x[n]$ before downsampling. Thus, if $x[n]$ is filtered by an ideal lowpass filter with cutoff frequency π/M, then the output $\tilde{x}[n]$ can be downsampled without aliasing, as illustrated in Figs. 3.19(d), (e), and (f). Note that the sequence $\tilde{x}_d[n] = \tilde{x}[nM]$ no longer represents the original underlying continuous-time signal $x_c(t)$. Rather $\tilde{x}_d[n] = \tilde{x}_c(nT')$, where $T' = MT$, and $\tilde{x}_c(t)$ is obtained from $x_c(t)$ by lowpass filtering with cutoff frequency $\Omega_c = \pi/T' = \pi/(MT)$.

From the preceding discussion we see that a general system for downsampling by a factor of M is the one shown in Fig. 3.20. Such a system is called a *decimator*, and downsampling by lowpass filtering followed by compression has been termed *decimation* (Crochiere and Rabiner, 1983).

3.6.2 Increasing the Sampling Rate by an Integer Factor

We have seen that reduction of the sampling rate of a discrete-time signal by an integer factor involves sampling the sequence in a manner analogous to sampling a continuous-time signal. Not surprisingly, increasing the sampling rate involves operations analogous to D/C conversion. To see this, consider a signal $x[n]$ whose sampling rate we wish to increase by a factor of L. If we consider the underlying continuous-time signal $x_c(t)$, the objective is to obtain samples

$$x_i[n] = x_c(nT'), \tag{3.79}$$

where $T' = T/L$, from the sequence of samples

$$x[n] = x_c(nT). \tag{3.80}$$

We will refer to the operation of increasing the sampling rate as *upsampling*.

It is clear from Eqs. (3.79) and (3.80) that

$$x_i[n] = x[n/L] = x_c(nT/L), \qquad n = 0, \pm L, \pm 2L, \ldots. \tag{3.81}$$

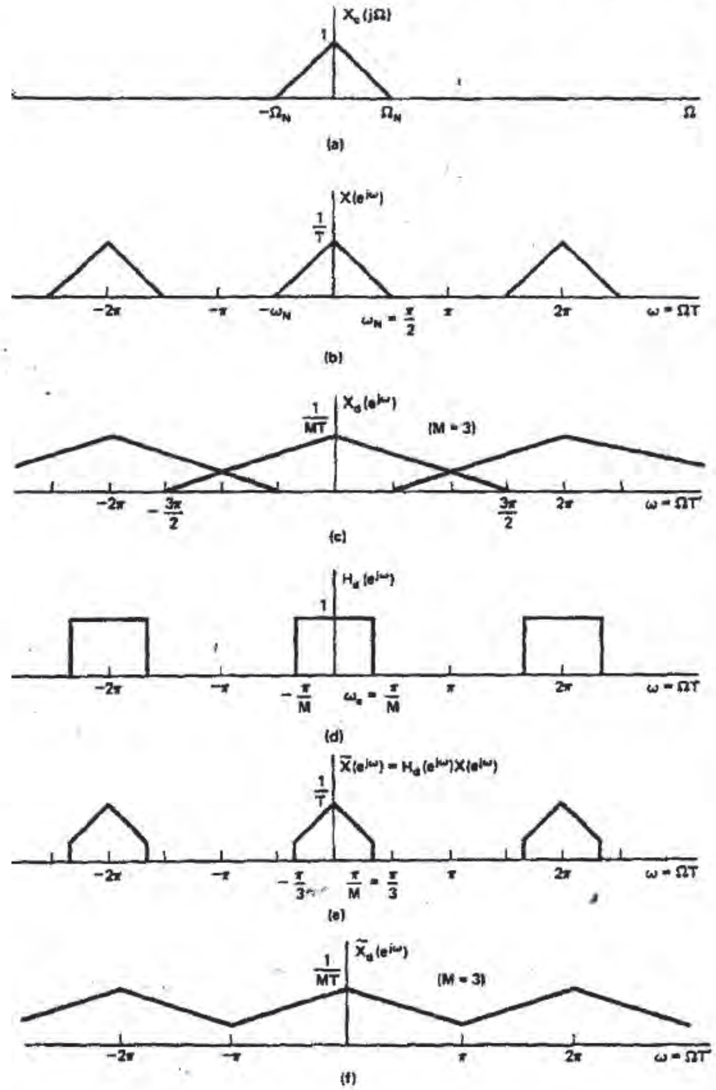


Figure 3.19 (a)–(c) Downsampling with aliasing. (d)–(f) Downsampling with prefiltering to avoid aliasing.

Figure 3.20 General system for sampling rate reduction by M.

Figure 3.21 shows a system for obtaining $x_i[n]$ from $x[n]$ using only discrete-time processing. The system on the left is called a *sampling rate expander* (Crochiere and Rabiner, 1983) or simply an *expander*. Its output is

$$x_e[n] = \begin{cases} x[n/L], & n = 0, \pm L, \pm 2L, \ldots, \\ 0, & \text{otherwise,} \end{cases} \tag{3.82}$$

or, equivalently,

$$x_e[n] = \sum_{k=-\infty}^{\infty} x[k]\delta[n-kL]. \tag{3.83}$$

The system on the right is a lowpass discrete-time filter with cutoff frequency π/L and gain L. This system plays a role similar to the ideal D/C converter in Fig. 3.19(b). First, we create a discrete-time impulse train $x_e[n]$, and then we lowpass filter to reconstruct the sequence.

The operation of the system in Fig. 3.21 is most easily understood in the frequency domain. The Fourier transform of $x_e[n]$ can be expressed as

$$\begin{aligned} X_e(e^{j\omega}) &= \sum_{n=-\infty}^{\infty}\left(\sum_{k=-\infty}^{\infty} x[k]\delta[n-kL]\right)e^{-j\omega n} \\ &= \sum_{k=-\infty}^{\infty} x[k]e^{-j\omega Lk} = X(e^{j\omega L}). \end{aligned} \tag{3.84}$$

Thus the Fourier transform of the output of the expander is a frequency-scaled version of the Fourier transform of the input; i.e., ω is replaced by ωL so that ω is now normalized by

$$\omega = \Omega T'. \tag{3.85}$$

This effect is illustrated in Fig. 3.22. Figure 3.22(a) shows a bandlimited continuous-time Fourier transform, and Fig. 3.22(b) shows the discrete-time Fourier transform of the sequence $x[n] = x_c(nT)$, where $\pi/T = \Omega_N$. Figure 3.22(c) shows $X_e(e^{j\omega})$ according to Eq. (3.84), with $L = 2$, and Fig. 3.22(e) shows the Fourier transform of the desired signal $x_i[n]$. We see that $X_i(e^{j\omega})$ can be obtained from $X_e(e^{j\omega})$ by correcting the amplitude scale from $1/T$ to $1/T'$ and by removing all the frequency-scaled images of $X_c(j\Omega)$ except at integer multiples of 2π. For the case depicted in Fig. 3.22 this requires a lowpass filter with a gain of 2 and cutoff frequency $\pi/2$, as shown in Fig.

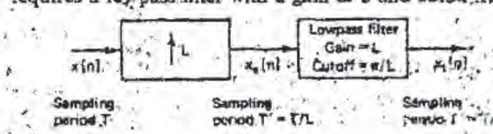


Figure 3.21 General system for sampling rate increase by L.

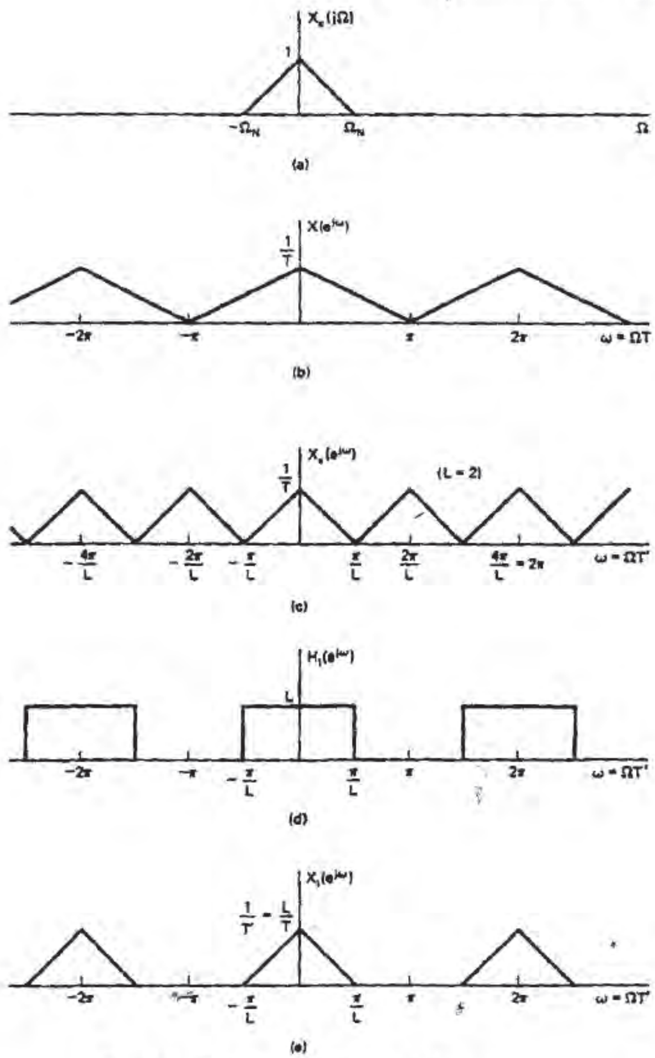


Figure 3.22 Frequency-domain illustration of interpolation.

3.22(d). In general, the required gain would be L, since $L(1/T) = [1/(T/L)] = 1/T'$, and the cutoff frequency would be π/L.

This example shows that the system of Fig. 3.21 does indeed give an output satisfying Eq. (3.79) if the input sequence $x[n] = x_c(nT)$ was obtained by sampling without aliasing. That system is therefore called an *interpolator* since it fills in the missing samples, and the operation of upsampling is therefore considered to be synonymous with *interpolation*.

As in the case of the D/C converter, it is possible to obtain an interpolation formula for $x_i[n]$ in terms of $x[n]$. First note that the impulse response of the lowpass filter in Fig. 3.21 is

$$h_i[n] = \frac{\sin(\pi n/L)}{\pi n/L}. \tag{3.86}$$

Using Eq. (3.83), we obtain

$$x_i[n] = \sum_{k=-\infty}^{\infty} x[k] \frac{\sin[\pi(n-kL)/L]}{\pi(n-kL)/L}. \tag{3.87}$$

The impulse response $h_i[n]$ has the properties

$$\begin{aligned} h_i[0] &= 1, \\ h_i[n] &= 0, \qquad n = \pm L, \pm 2L, \ldots. \end{aligned} \tag{3.88}$$

Thus for the ideal lowpass interpolation filter we have

$$x_i[n] = x[n/L] = x_c(nT/L) = x_c(nT'), \qquad n = 0, \pm L, \pm 2L, \ldots, \tag{3.89}$$

as desired. The fact that $x_i[n] = x_c(nT')$ for all n follows from our frequency-domain argument.

In practice, ideal lowpass filters cannot be implemented exactly, but we will see in Chapter 7 that very good approximations can be designed. (Also see Schafer and Rabiner, 1973, and Oetken et al., 1975.) In some cases very simple interpolation procedures are adequate. Since linear interpolation is often used, although generally it is not very accurate, it is worthwhile to examine linear interpolation in the present context.

Linear interpolation can be accomplished by the system of Fig. 3.21 if the filter has impulse response

$$h_{lin}[n] = \begin{cases} 1 - |n|/L, & |n| \le L, \\ 0, & \text{otherwise}, \end{cases} \tag{3.90}$$

as shown in Fig. 3.23 for $L = 5$. With this filter, the interpolated output will be

$$x_{lin}[n] = \sum_{k=-\infty}^{\infty} x_e[k] h_{lin}[n-k] = \sum_{k=-\infty}^{\infty} x[k] h_{lin}[n-kL]. \tag{3.91}$$

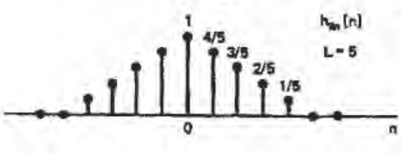


Figure 3.23 Impulse response for linear interpolation.

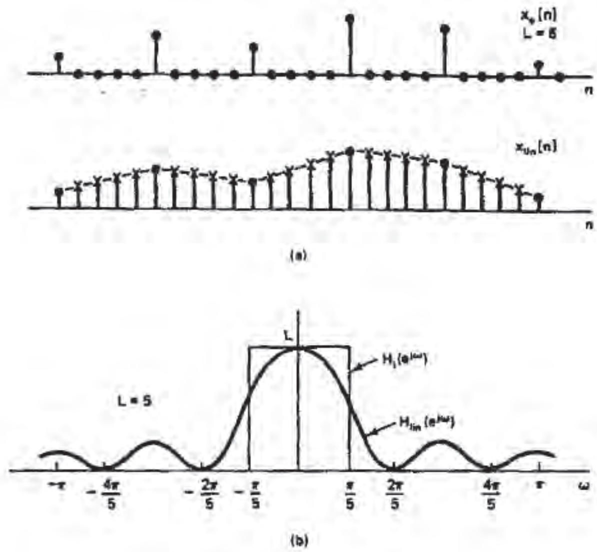


Figure 3.24 (a) Illustration of linear interpolation by filtering. (b) Frequency response of linear interpolator compared with ideal lowpass interpolation filter.

Figure 3.24(a) depicts $x_e[n]$ and $x_{lin}[n]$ for the case $L = 5$. From this figure we see that $x_{lin}[n]$ is identical to the sequence obtained by linear interpolation between the samples. Note that

$$\begin{aligned} h_{lin}[0] &= 1, \\ h_{lin}[n] &= 0, \qquad n = \pm L, \pm 2L, \ldots, \end{aligned} \tag{3.92}$$

so that

$$x_{lin}[n] = x[n/L] \quad \text{at} \quad n = 0, \pm L, \pm 2L, \ldots. \tag{3.93}$$

The amount of distortion in the intervening samples can be gauged by comparing the frequency response of the linear interpolator to the ideal lowpass interpolator for a factor-of-L interpolation. It can be shown (see Problem 3.22) that

$$H_{lin}(e^{j\omega}) = \frac{1}{L}\left[\frac{\sin(\omega L/2)}{\sin(\omega/2)}\right]^2. \tag{3.94}$$

This function is plotted in Fig. 3.24(b) for $L = 5$ together with the ideal lowpass interpolation filter. It is clear from this figure that if the original signal is sampled at the Nyquist rate, linear interpolation will not be very good since the output of the filter will contain considerable energy in the band $\pi/L < |\omega| \leq \pi$. However, if the original sampling rate is much higher than the Nyquist rate, then the linear interpolator will be more successful in removing the frequency-scaled images of $X_c(j\Omega)$ at multiples of $2\pi/L$. This is intuitively reasonable since if the original sampling rate greatly exceeds the Nyquist rate, the signal will not vary significantly between samples, and thus linear interpolation should be reasonably accurate.

3.6.3 Changing the Sampling Rate by a Noninteger Factor

We have shown how to increase or decrease the sampling rate of a sequence by an integer factor. By combining decimation and interpolation it is possible to change the sampling rate by a noninteger factor. Specifically, consider Fig. 3.25(a), which shows an interpolator that decreases the sampling period from T to T/L, followed by a decimator, which increases the sampling period by M, producing an output sequence $\tilde{x}_d[n]$ that has an effective sampling period of $T' = TM/L$. By choosing L and M appropriately, we can approach arbitrarily close to any desired ratio of sampling periods. For example, if $L = 100$ and $M = 101$, then $T' = 1.01T$.

If $M > L$ there is a net increase in the sampling period (decrease in sampling rate) and if $M < L$, the opposite is true. From Fig. 3.25(a) we see that the interpolation and decimation filters are in cascade, so that they can be combined as shown in Fig. 3.25(b) into one lowpass filter with gain L and cutoff equal to the minimum of π/L and π/M. If $M > L$, then π/M is the dominant cutoff frequency and there is a net reduction in sampling rate. As pointed out in Section 3.6.1, if $x[n]$ was obtained by sampling at the Nyquist rate, the sequence $\tilde{x}_d[n]$ will represent a lowpass-filtered version of the original underlying bandlimited signal if we are to

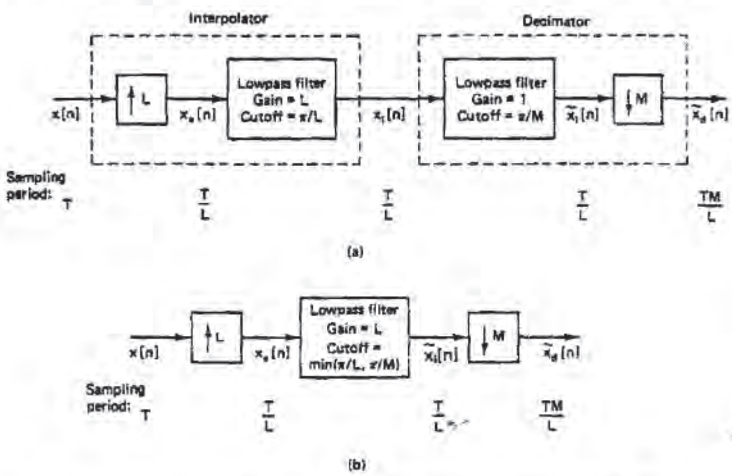


Figure 3.25 (a) System for changing the sampling rate by a noninteger factor. (b) Simplified system in which the decimation and interpolation filters are combined.

avoid aliasing. On the other hand, if $M < L$, then π/L is the dominant cutoff frequency and there will be no need to further limit the bandwidth of the signal below the original Nyquist frequency.

3.7 PRACTICAL CONSIDERATIONS

So far in this chapter our discussions of the representation of continuous-time signals by discrete-time signals have focused on idealized models of periodic sampling and bandlimited interpolation. We have formalized our discussions in terms of an idealized sampling system that we have called the *ideal continuous-to-discrete (C/D) converter* and an idealized bandlimited interpolator system called the *ideal discrete-to-continuous (D/C) converter*. These idealized conversion systems allow us to concentrate on the essential mathematical details of the relationship between a bandlimited signal and its samples. For example, in Section 3.4 we used the idealized C/D and D/C conversion systems to show that linear time-invariant discrete-time systems can be used in the configuration of Fig. 3.26(a) to implement linear time-invariant continuous-time systems if the input is bandlimited and the sampling rate exceeds the Nyquist rate. In a practical setting, continuous-time signals are not precisely bandlimited, ideal filters cannot be realized, and the ideal C/D and D/C converters can only be approximated. The block diagram of Fig. 3.26(b) shows a more realistic model for digital processing of continuous-time (analog) signals. In this section we will examine some of the imperfections introduced by each of the components of the system in Fig. 3.26(b). While it is difficult to carry out an exact analysis, our approximate analysis will indicate the effects of each imperfection and suggest that the system of Fig. 3.26(b) can very closely approximate the system of Fig. 3.26(a).

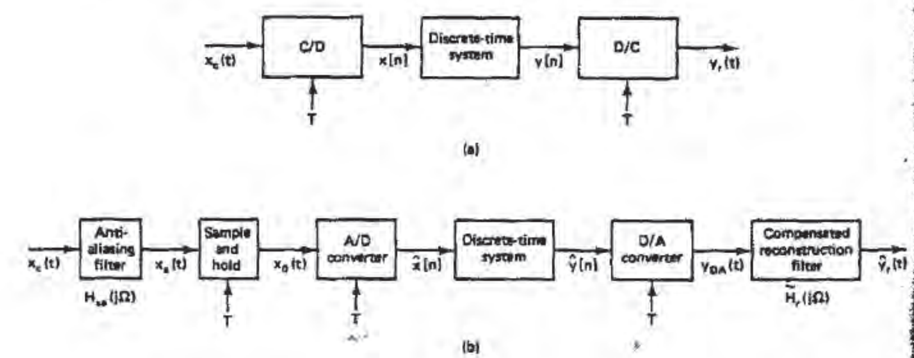


Figure 3.26 (a) Discrete-time filtering of continuous-time signals. (b) A more realistic model.

PT00600U

EXPRESS MAIL - MAILING LABEL NUMBER EG938802607US

DATE OF DEPOSIT February 28, 1995

I HEREBY CERTIFY THAT THIS CORRESPONDENCE IS BEING DEPOSITED WITH THE UNITED STATES POSTAL SERVICE EXPRESS MAIL POST OFFICE TO ADDRESSEE ON THE DATE INDICATED ABOVE AND IS ADDRESSED TO THE COMMISSIONER OF PATENTS AND TRADEMARKS, WASHINGTON, D.C. 20231.

Cathy L. Tredinnick
Printed name of person mailing paper or fee

Cathy L. Tredinnick
Signature of person mailing paper or fee

PT00600U

EXPRESS MAIL - MAILING LABEL NUMBER EG938802607US

DATE OF DEPOSIT February 28, 1995

I HEREBY CERTIFY THAT THIS CORRESPONDENCE IS BEING DEPOSITED WITH THE UNITED STATES POSTAL SERVICE EXPRESS MAIL POST OFFICE TO ADDRESSEE ON THE DATE INDICATED ABOVE AND IS ADDRESSED TO THE COMMISSIONER OF PATENTS AND TRADEMARKS, WASHINGTON, D.C. 20231.

Cathy L. Tredinnick

Printed name of person mailing paper or fee

Cathy L. Tredinnick

Signature of person mailing paper or fee

MAIL ROOM
82 APR 3 1995
PAT. & TRADEMARK OFF.

PT00600U #3
Prior Art
aw 6/14/95

UNITED STATES PATENT AND TRADEMARK OFFICE

APPLICANTS: Leitch, et al.

SERIAL NO.: 08/395,747

FILED: February 28, 1995

EXAMINER:

ART UNIT: 2308

CASE NO.: PT00600U

RECEIVED
APR 07 1995
GROUP 2300

APPLICATION: Voice Compression Method And Apparatus In A Communication System

---

Motorola, Inc.
Patent Department
1500 Gateway Blvd.
Boynton Beach, FL 33426
March 31, 1995

DISCLOSURE STATEMENT UNDER 37 CFR §1.99

Hon. Commissioner of Patents
and Trademarks
Washington, DC 20231

Sir:

Under 37 CFR §1.99, a further list of documents that has come to the Applicants' attention is disclosed on the attached form PTO-1449 that may be material to the examination of this application. A copy of each of the documents is included herewith for the Examiner's consideration.

(1) The information included herewith 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 statement, or

(2) The information included herewith was not cited in a communication from a foreign patent office in a counterpart foreign application or, to the knowledge of the person signing the certification after making reasonable inquiry, was known to any individual designated in §1.56(c) more than three months prior to the filing of this statement.

PT00600U

The following comments are directed to the documents disclosed in the accompanying form PTO-1449:

PATENTS

U.S. Patent No. 4,875,038 issued October 17, 1989 to Siwiak, et al.
U.S. Patent No. 4,882,579 issued November 21, 1989 to Siwiak, et al.

Respectfully submitted,

Pablo Meles

Pablo Meles
Attorney for Applicants
Registration No.: 33,308
Phone: (407) 739-2860
Fax: (407) 739-2825

#3

FORM PTO-1449 U.S. DEPARTMENT OF COMMERCE, PATENT AND TRADEMARK OFFICE

LIST OF ART CITED BY APPLICANT

(Use several sheets if necessary)

RECEIVED
APR 0 7 1995
GROUP 2300

ATTY. DOCKET NO.	SERIAL NO.	FILING DATE
PT00600U	08/396,747	February 28, 1995
APPLICANT		**GROUP**
LEITCH, et al.		2308

U.S. PATENT DOCUMENTS

EXAMINER INITIAL		DOCUMENT NUMBER	DATE	NAME	CLASS	SUBCLASS	FILING DATE IF APPROPRIATE
C	AA	4,882,579	11/21/89	Siwiak	340	825.44	
C	AB	4,875,038	10/17/89	Siwiak, et al.	340	825.44	
	AC						
	AD						
	AE						
	AF						
	AG						
	AH						
	AI						
	AJ						
	AK						

FOREIGN PATENT DOCUMENTS

		DOCUMENT NUMBER	DATE	COUNTRY	CLASS	SUBCLASS	TRANSLATION YES	TRANSLATION NO.
	AL							
	AM							
	AN							
	AO							

OTHER PRIOR ART *(Including Author, Title, Date, Pertinent Pages, Etc.)*

	AP	
	AR	
	AS	
	AT	

EXAMINER	DATE CONSIDERED
Demetra Smith	9/11/96

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

Notice of Allowability	Application No. 08/395,747	Applicant(s) Leitch et al.
	Examiner Demetra R. Smith (9/13/96)	Group Art Unit 2414

All claims being allowable, PROSECUTION ON THE MERITS IS (OR REMAINS) CLOSED in this application. If not included herewith (or previously mailed), a Notice of Allowance and Issue Fee Due or other appropriate communication will be mailed in due course.

☒ This communication is responsive to *Feb 28, 1995*.

☒ The allowed claim(s) is/are *1-24*.

☐ The drawings filed on __________ are acceptable.

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

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

☐ received.

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

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

*Certified copies not received: __________.

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

A SHORTENED STATUTORY PERIOD FOR RESPONSE to comply with the requirements noted below is set to EXPIRE **THREE MONTHS** FROM THE "DATE MAILED" of this Office action. Failure to timely comply will result in ABANDONMENT of this application. Extensions of time may be obtained under the provisions of 37 CFR 1.136(a).

☐ Note the attached EXAMINER'S AMENDMENT or NOTICE OF INFORMAL APPLICATION, PTO-152, which discloses that the oath or declaration is deficient. A SUBSTITUTE OATH OR DECLARATION IS REQUIRED.

☐ Applicant MUST submit NEW FORMAL DRAWINGS

☐ because the originally filed drawings were declared by applicant to be informal.

☐ including changes required by the Notice of Draftsperson's Patent Drawing Review, PTO-948, attached hereto or to Paper No. ____.

☐ including changes required by the proposed drawing correction filed on __________, which has been approved by the examiner.

☐ including changes required by the attached Examiner's Amendment/Comment.

Identifying indicia such as the application number (see 37 CFR 1.84(c)) should be written on the reverse side of the drawings. The drawings should be filed as a separate paper with a transmittal lettter addressed to the Official Draftsperson.

☐ Note the attached Examiner's comment regarding REQUIREMENT FOR THE DEPOSIT OF BIOLOGICAL MATERIAL.

Any response to this letter should include, in the upper right hand corner, the APPLICATION NUMBER (SERIES CODE/SERIAL NUMBER). If applicant has received a Notice of Allowance and Issue Fee Due, the ISSUE BATCH NUMBER and DATE of the NOTICE OF ALLOWANCE should also be included.

Attachment(s)

☒ Notice of References Cited, PTO-892

☒ Information Disclosure Statement(s), PTO-1449, Paper No(s). *2 and 3*

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

☐ Notice of Informal Patent Application, PTO-152

☐ Interview Summary, PTO-413

☐ Examiner's Amendment/Comment

☐ Examiner's Comment Regarding Requirement for Deposit of Biological Material

☒ Examiner's Statement of Reasons for Allowance

DETAILED ACTION

Allowable Subject Matter

1. Claims 1-24 are allowed.

2. The following is an examiner's statement of reasons for allowance: The prior art does not show separately or in combination the limitations of subchanneling the voice communication resource and simultaneously placing at least one of each of the plurality of voice signals on a subchannel; compressing the time of each of the voice signals within each of the subchannels; wherein the step of compressing the time of each of the voice signals further comprises the step of using Waveform Similarity based Overlap-Add (WSOLA) time compression on the voice signals; compressing the bandwidth of plurality of voice signals by subchanneling the voice communication resource and modulating each of the plurality of voice signals about a plurality of pilot signals within the voice communication resource; a processing device for demodulating the received processed signal using single side band demodulation and time-scale expansion to provide a reconstructed signal; subchannelizing the communication resource into a predetermined number of subchannels; and a digital signal processor for performing single sideband demodulation and at least one of the functions of filtering a pilot carrier, performing

automatic gain control using a feedforward loop, or decompanding of the digitized received signal to provide a processed signal.

Any comments considered necessary by applicant must be submitted no later than the payment of the issue fee and, to avoid processing delays, should preferably accompany the issue fee. Such submissions should be clearly labeled "Comments on Statement of Reasons for Allowance."

Conclusion

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

Dejmek et al. ('898) dislcoses a method for time compressing a voice message in a voice messaging method for selective call receivers.

Hieatt, III ('215) discloses s selective call transceiver unit, selective call base station in which the receiver processes the radio signals and produces output demodulated information. The demodulated information is coupled into the input of a microprocessor which processed the information and incorporates conventional analog-to-digital and digital-to-voice message from the receiver.

Liberti, Jr. et al. discloses a selective call base station for use in a selective call communication system, comprising a transmitter coupled to the controller for transmitting selective call signals to selective call receivers operating in the selective call communication system, a receiver coupled to the controller for receiving a modulated radio signal from a transmitter within one of the selective call receivers, an analog-to-digital converter for digitizing the modulated radio signal and a symbol demodulator coupled to the carrier frequency estimator for demodulating a symbol value corresponding to one of the serially transmitted symbol intervals.

Orlen et al. ('981) discloses a selective call communication system that includes an encoder for encoding and compressing the data to form compressed data, and a controller for transferring the compressed data to a selective call terminal.

Puckette ('589) discloses a digitally implemented single sideband (SSB) transmitter simultaneously generates a desired signaling waveform in response to a single data signal input, multiplies it by a carrier function, generates the Hilbert transform of the signaling waveform and multiplies it by a quadrature carrier function, and sums the two products to produce the SSB output.

Wong ('174) discloses an analog signal stacking method and apparatus permits transmission of n analog signals of given spectrum width over a narrow band communication circuit having bandwidth approximately equal to the spectrum width of one analog signal by compressing the spectrum of each signal by a factor of 1/n.

4. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Demetra R. Smith whose telephone number is (703) 308-6989. The examiner can normally be reached on Monday-Friday from 8:00 to 5:00.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, E. Todd Voeltz, can be reached on (703) 305-9714. The fax phone number for this Group is (703) 305-9731.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the Group receptionist whose telephone number is (703) 305-3800.

Demetra R. Smith

September 15, 1996

EMANUEL T. VOELTZ
SUPERVISORY PATENT EXAMINER
GROUP 2400

08/764656

Notice of References Cited	Application No. 08/~~395,747~~	Applicant(s) Leitch et al.	
	Examiner Demetra R. Smith (9/13/96)	Group Art Unit 2414	Page 1 of 1

U.S. PATENT DOCUMENTS

*		DOCUMENT NO.	DATE	NAME	CLASS	SUBCLASS
x	A	3,793,589	2-19-74	Puckette	325	137
x	B	4,586,174	4-29-86	Wong	370	69.1
x	C	5,533,062	7-2-96	Liberti, Jr. et al	375	334
x	D	5,068,898	11-26-91	Dejmek et al.	381	29
x	E	5,387,981	2-7-96	Orlen et al.	358	400
x	F	5,535,215	7-9-96	Hieatt, III	370	95.1
	G					
	H					
	I					
	J					
	K					
	L					
	M					

FOREIGN PATENT DOCUMENTS

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

NON-PATENT DOCUMENTS

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

Demetra R. Smith

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

U. S. Patent and Trademark Office
PTO-892 (Rev. 9-95)

Notice of References Cited

Part of Paper No. 4

Form PTO 948 (Rev. 10-94) U.S. DEPARTMENT OF COMMERCE - Patent and Trademark Office Application No. 395-747

NOTICE OF DRAFTSPERSON'S PATENT DRAWING REVIEW

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

The drawings filed (insert date) 2/28/97 are
A. ✓ not objected to by the Draftsperson under 37 CFR 1.84 or 1.152.
B. ___ objected to by the Draftsperson under 37 CFR 1.84 or 1.152 as indicated below. The Examiner will require submission of new, corrected drawings when necessary. Corrected drawings must be submitted according to the instructions on the back of this Notice.

1. DRAWINGS. 37 CFR 1.84(a): Acceptable categories of drawings: Black ink. Color.
   - ___ Not black solid lines. Fig(s) ___
   - ___ Color drawings are not acceptable until petition is granted. Fig(s) ___
2. PHOTOGRAPHS. 37 CFR 1.84(b)
   - ___ Photographs are not acceptable until petition is granted. Fig(s) ___
   - ___ Photographs not properly mounted (must use brystol board or photographic double-weight paper). Fig(s) ___
   - ___ Poor quality (half-tone). Fig(s) ___
3. GRAPHIC FORMS. 37 CFR 1.84(d)
   - ___ Chemical or mathematical formula not labeled as separate figure. Fig(s) ___
   - ___ Group of waveforms not presented as a single figure, using common vertical axis with time extending along horizontal axis. Fig(s) ___
   - ___ Individuals waveform not identified with a separate letter designation adjacent to the vertical axis. Fig(s) ___
4. TYPE OF PAPER. 37 CFR 1.84(e)
   - ___ Paper not flexible, strong, white, smooth, nonshiny, and durable. Sheet(s) ___
   - ___ Erasures, alterations, overwritings, interlineations, cracks, creases, and folds copy machine marks not accepted. Fig(s) ___
   - ___ Mylar, velum paper is not acceptable (too thin). Fig(s) ___
5. SIZE OF PAPER. 37 CFR 1.84(f): Acceptable sizes:
   - 21.6 cm. by 35.6 cm. (8 1/2 by 14 inches)
   - 21.6 cm. by 33.1 cm. (8 1/2 by 13 inches)
   - 21.6 cm. by 27.9 cm. (8 1/2 by 11 inches)
   - 21.0 cm. by 29.7 cm. (DIN size A4)
   - ___ All drawing sheets not the same size. Sheet(s) ___
   - ___ Drawing sheet not an acceptable size. Sheet(s) ___
6. MARGINS. 37 CFR 1.84(g): Acceptable margins:

Paper size

	21.6 cm. X 35.6 cm. (8 1/2 X 14 inches)	21.6 cm. X 33.1 cm. (8 1/2 X 13 inches)	21.6 cm. X 27.9 cm. (8 1/2 X 11 inches)	21.0 cm. X 29.7 cm. (DIN Size A4)
T	5.1 cm. (2")	2.5 cm. (1")	2.5 cm. (1")	2.5 cm.
L	.64 cm. (1/4")	.64 cm. (1/4")	.64 cm. (1/4")	2.5 cm.
R	.64 cm. (1/4")	.64 cm. (1/4")	.64 cm. (1/4")	1.5 cm.
B	.64 cm. (1/4")	.64 cm. (1/4")	.64 cm. (1/4")	1.0 cm.

___ Margins do not conform to chart above. Sheet(s) ___
___ Top (T) ___ Left (L) ___ Right (R) ___ Bottom (B)

7. VIEWS. 37 CFR 1.84(h)
   REMINDER: Specification may require revision to correspond to drawing changes.
   - ___ All views not grouped together. Fig(s) ___
   - ___ Views connected by projection lines or lead lines. Fig(s) ___
   - Partial views. 37 CFR 1.84(h) 2
   - ___ View and enlarged view not labled separatly or properly. Fig(s) ___
   - Sectional views. 37 CFR 1.84 (h) 3
   - ___ Hatching not indicated for sectional portions of an object. Fig(s) ___
   - ___ Cross section not drawn same as view with parts in cross section with regularly spaced parallel oblique strokes. Fig(s) ___
8. ARRANGEMENT OF VIEWS. 37 CFR 1.84(i)
   - ___ Words do not appear on a horizontal, left-to-right fashion when page is either upright or turned so that the top becomes the right side, except for graphs. Fig(s) ___
9. SCALE. 37 CFR 1.84(k)
   - ___ Scale not large enough to show mechanism with crowding when drawing is reduced in size to two thirds in reproduction. Fig(s) ___
   - ___ Indication such as "actual size" or scale 1/2" not permitted. Fig(s) ___
10. CHARACTER OF LINES, NUMBERS, & LETTERS. 37 CFR 1.84(l)
   - ___ Lines, numbers & letters not uniformly thick and well defined, clean, durable, and black (except for color drawings). Fig(s) ___
11. SHADING. 37 CFR 1.84(m)
   - ___ Solid black shading areas not permitted. Fig(s) ___
   - ___ Shade lines, pale, rough and blurred. Fig(s) ___
12. NUMBERS, LETTERS, & REFERENCE CHARACTERS. 37 CFR 1.84(p)
   - ___ Numbers and reference characters not plain and legible. 37 CFR 1.84(p)(1) Fig(s) ___
   - ___ Numbers and reference characters not oriented in same direction as the view. 37 CFR 1.84(p)(1) Fig(s) ___
   - ___ English alphabet not used. 37 CFR 1.84(p)(2) Fig(s) ___
   - ___ Numbers, letters, and reference characters do not measure at least .32 cm. (1/8 inch) in height. 37 CFR(p)(3) Fig(s) ___
13. LEAD LINES. 37 CFR 1.84(q)
   - ___ Lead lines cross each other. Fig(s) ___
   - ___ Lead lines missing. Fig(s) ___
14. NUMBERING OF SHEETS OF DRAWINGS. 37 CFR 1.84(t)
   - ___ Sheets not numbered consecutively, and in Arabic numerals, beginning with number 1. Sheet(s) ___
15. NUMBER OF VIEWS. 37 CFR 1.84(u)
   - ___ Views not numbered consecutively, and in Arabic numerals, beginning with number 1. Fig(s) ___
   - ___ View numbers not preceded by the abbreviation Fig. Fig(s) ___
16. CORRECTIONS. 37 CFR 1.84(w)
   - ___ Corrections not made from prior PTO-948. Fig(s) ___
17. DESIGN DRAWING. 37 CFR 1.152
   - ___ Surface shading shown not appropriate. Fig(s) ___
   - ___ Solid black shading not used for color contrast. Fig(s) ___

COMMENTS:

2/18/97

UNITED STATES DEPARTMENT OF COMMERCE
Patent and Trademark Office

Address: Box ISSUE FEE
ASSISTANT COMMISSIONER FOR PATENTS
Washington, D.C. 20231

NOTICE OF ALLOWANCE AND ISSUE FEE DUE

2431/1901

JOHN H. MOORE
MOTOROLA INC
1500 GATEWAY BLVD MS 96
BOYNTON BEACH FL 33426-8292

APPLICATION NO.	FILING DATE	TOTAL CLAIMS	EXAMINER AND GROUP ART UNIT		DATE MAILED
08/395,747	02/28/95	024	SMITH, D	2414	10/01/96

First Named Applicant: LEITCH, CLIFFORD D.

TITLE OF INVENTION: VOICE COMPRESSION METHOD AND APPARATUS IN A COMMUNICATION SYSTEM

ATTY'S DOCKET NO.	CLASS-SUBCLASS	BATCH NO.	APPLN. TYPE	SMALL ENTITY	FEE DUE	DATE DUE
S PD04630U	364-514.00R	P03	UTILITY	NO	$1290.00	01/02/97

THE APPLICATION IDENTIFIED ABOVE HAS BEEN EXAMINED AND IS ALLOWED FOR ISSUANCE AS A PATENT. PROSECUTION ON THE MERITS IS CLOSED.

THE ISSUE FEE MUST BE PAID WITHIN THREE MONTHS FROM THE MAILING DATE OF THIS NOTICE OR THIS APPLICATION SHALL BE REGARDED AS ABANDONED. THIS STATUTORY PERIOD CANNOT BE EXTENDED.

HOW TO RESPOND TO THIS NOTICE:

I. Review the SMALL ENTITY status shown above.
If the SMALL ENTITY is shown as YES, verify your current SMALL ENTITY status:

A. If the status is changed, pay twice the amount of the FEE DUE shown above and notify the Patent and Trademark Office of the change in status, or
B. If the status is the same, pay the FEE DUE shown above.

If the SMALL ENTITY is shown as NO:

A. Pay FEE DUE shown above, or

B. File verified statement of Small Entity Status before, or with, payment of 1/2 the FEE DUE shown above.

II. Part B of this notice should be completed and returned to the Patent and Trademark Office (PTO) with your ISSUE FEE. Even if the ISSUE FEE has already been paid by charge to deposit account, Part B should be completed and returned. If you are charging the ISSUE FEE to your deposit account, section "6b" of Part B should be completed.

III. All communications regarding this application must give application number and batch number. Please direct all communication prior to issuance to Box ISSUE FEE unless advised to the contrary.

IMPORTANT REMINDER: 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.

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

SERIAL NUMBER	FILING DATE	FIRST NAMED APPLICANT	ATTORNEY DOCKET NO.
08/395,747	02/28/95	LEITCH	C PT00600U

7512/0317

JOHN H. MOORE
MOTOROLA INC
1500 GATEWAY BLVD MS 96
BOYNTON BEACH FL 33426-8292

EXAMINER	
SMITH, P	
ART UNIT	PAPER NUMBER
2414	5

DATE MAILED: 03/17/97

NOTICE OF ABANDONMENT

This application is abandoned in view of:

1. ☐ Applicant's failure to respond to the Office letter, mailed __________.
2. ☐ Applicant's letter of express abandonment which is in compliance with 37 C.F.R. 1.138.
3. ☐ Applicant's failure to timely file the response received __________ within the period set in the Office letter.
4. ☐ Applicant's failure to pay the required issue fee within the statutory period of 3 months from the mailing date of __________ of the Notice of Allowance.

   ☐ The issue fee was received on __________.

   ☐ The issue fee has not been received in Allowed Files Branch as of __________.

   In accordance with 35 U.S.C. 151, and under the provisions of 37 C.F.R. 1.316(b), applicant(s) may petition the Commissioner to accept the delayed payment of the issue fee if the delay in payment was unavoidable. The petition must be accompanied by the issue fee, unless it has been previously submitted, in the amount specified by 37 C.F.R. 1.17(l), and a verified showing as to the causes of the delay.

   If applicant(s) never received the Notice of Allowance, a petition for a new Notice of Allowance and withdrawal of the holding of abandonment may be appropriate in view of Delgar Inc. v. Schuyler, 172 U.S.P.Q. 513.

5. ☐ Applicant's failure to timely correct the drawings and/or submit new or substitute formal drawings by __________ as required in the last Office action.

   ☐ The corrected and/or substitute drawings were received on __________.

6. ☑ The reason(s) below.

Abandoned in view of file
wrapper continuation
application filed under
37 CFR 1.62

PTO-1432 (Rev. 4/93)

PATENT APPLICATION FEE DETERMINATION RECORD

Effective October 1, 1994

Application or Docket Number: 08/395747

CLAIMS AS FILED - PART I

FOR	NUMBER FILED (Column 1)		NUMBER EXTRA (Column 2)	SMALL ENTITY RATE	SMALL ENTITY FEE	OR	OTHER THAN SMALL ENTITY RATE	OTHER THAN SMALL ENTITY FEE
BASIC FEE					365.00	OR		730.00
TOTAL CLAIMS	24	minus 20 =	* 4	x$11=		OR	x$22=	88
INDEPENDENT CLAIMS	7	minus 3 =	* 4	x38=		OR	x76=	304
MULTIPLE DEPENDENT CLAIM PRESENT				+120=		OR	+240=	
				TOTAL		OR	TOTAL	1122

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

CLAIMS AS AMENDED - PART II

AMENDMENT A

	CLAIMS REMAINING AFTER AMENDMENT (Column 1)		HIGHEST NUMBER PREVIOUSLY PAID FOR (Column 2)	PRESENT EXTRA (Column 3)	SMALL ENTITY RATE	ADDITIONAL FEE	OR	OTHER THAN SMALL ENTITY RATE	ADDITIONAL FEE
Total	*	Minus	**	=	x$11=		OR	x$22=	
Independent	*	Minus	***	=	x38=		OR	x76=	
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM					+120=		OR	+240=	
					TOTAL ADDIT. FEE		OR	TOTAL ADDIT. FEE	

AMENDMENT B

	CLAIMS REMAINING AFTER AMENDMENT (Column 1)		HIGHEST NUMBER PREVIOUSLY PAID FOR (Column 2)	PRESENT EXTRA (Column 3)	SMALL ENTITY RATE	ADDITIONAL FEE	OR	OTHER THAN SMALL ENTITY RATE	ADDITIONAL FEE
Total	*	Minus	**	=	x$11=		OR	x$22=	
Independent	*	Minus	***	=	x38=		OR	x76=	
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM					+120=		OR	+240=	
					TOTAL ADDIT. FEE		OR	TOTAL ADDIT. FEE	

AMENDMENT C

	CLAIMS REMAINING AFTER AMENDMENT (Column 1)		HIGHEST NUMBER PREVIOUSLY PAID FOR (Column 2)	PRESENT EXTRA (Column 3)	SMALL ENTITY RATE	ADDITIONAL FEE	OR	OTHER THAN SMALL ENTITY RATE	ADDITIONAL FEE
Total	*	Minus	**	=	x$11=		OR	x$22=	
Independent	*	Minus	***	=	x38=		OR	x76=	
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM					+120=		OR	+240=	
					TOTAL ADDIT. FEE		OR	TOTAL ADDIT. FEE	

* If the entry in column 1 is less than the entry in column 2, write "0" in column 3.
** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 20, enter "20."
*** If the Highest Number Previously Paid For" IN THIS SPACE is less than 3, enter "3."
The Highest Number Previously Paid For" (Total or Independent) is the highest number found in the appropriate box in column 1.

FORM PTO-875 (Rev. 10/94)

Patent and Trademark Office, U.S. DEPARTMENT OF COMMERCE

BAR CODE LABEL	U.S. PATENT APPLICATION		
SERIAL NUMBER 08/764,656	FILING DATE 12/11/96	CLASS 364	GROUP ART UNIT 2414

APPLICANT

CLIFFORD D. LEITCH, CORAL SPRINGS, FL; ROBERT J. SCHWENDEMAN, POMPANO BEACH, FL; KAZIMIERZ SIWIAK, CORAL SPRINGS, FL; WILLIAM J. KUZNICKI, CORAL SPRINGS, FL; SUNIL SATYAMURTI, DELRAY BEACH, FL.

CONTINUING DATA*******************
VERIFIED THIS APPLN IS A CON OF 08/395,747 02/28/95

FOREIGN/PCT APPLICATIONS**********
VERIFIED

FOREIGN FILING LICENSE GRANTED 02/20/97

STATE OR COUNTRY	SHEETS DRAWING	TOTAL CLAIMS	INDEPENDENT CLAIMS	FILING FEE RECEIVED	ATTORNEY DOCKET NO.
FL	14	24	7	$1,178.00	PT00600UC01

ADDRESS

MOTOROLA INC
IP LAW DEPARTMENT MS 96
1500 GATEWAY BLVD
BOYNTON BEACH FL 33426-8753

TITLE

VOICE COMPRESSION METHOD AND APPARATUS IN A COMMUNICATION SYSTEM

This is to certify that annexed hereto is a true copy from the records of the United States Patent and Trademark Office of the application which is identified above.

By authority of the
COMMISSIONER OF PATENTS AND TRADEMARKS

Date Certifying Officer

PATENT APPLICATION SERIAL NO. 08/764656

U.S. DEPARTMENT OF COMMERCE
PATENT AND TRADEMARK OFFICE
FEE RECORD SHEET

ok
22

RT18104 01/29/97 08764656 13-4778 180 101 22.00CR

PTO-1556
(5/87)

08 54656

Affwc #6 /Pref m.c. 3-18-97

UNITED STATES PATENT AND TRADEMARK OFFICE

Docket No.: PT00600UC01

Date: December 3, 1996

Box FWC
Commissioner of Patents
and Trademarks
Washington, DC 20231

Sir:

This is a request for filing a:

__X__ Continuation file wrapper continuing application under 37 CFR 1.62, of pending prior application Serial No. 08/395,747 filed on February 28, 1995 by Leitch, et al. for "VOICE COMPRESSION METHOD AND APPARATUS IN A COMMUNICATION SYSTEM."

1. __X__ The filing fee is calculated below:

CLAIMS REMAINING AFTER ENTRY OF ANY REQUESTED RULE 116 AND/OR PRELIMINARY AMENDMENTS

FOR	NUMBER FILED	NUMBER EXTRA	RATE	BASIC FEE $770
TOTAL CLAIMS	24 - 20 =	4	X $ 22	= $110
INDEP. CLAIMS	7 - 3 =	4	X $ 80	= $320
	TOTAL FILING FEE			$1200

2. __X___ The Commissioner is hereby authorized to charge any application filing fees which may be required, or credit any overpayment to Deposit Account No. 13-4778. Three originals of this sheet are enclosed.

3. _____ A check in the amount of $____ is enclosed.

4. _____ Cancel claims ________________________.

5. __X___ Amend the specification by inserting before the first line the sentence: --This is a continuation of application Serial No. 08/395,747 filed February 28, 1995, now abandoned.

A1 3/12/97

48

6. __X___ Transfer the contents of the prior application and its file wrapper to this application and abandon said prior application. Three originals of this sheet are enclosed for filing in the prior application file.

7. _____ Please enter the amendment dated __________ which was filed in the prior application under 37 CFR 1.116, but not entered.

8. _____ A preliminary amendment is enclosed presenting additional claims and/or changes to be entered in the new application after a filing date has been granted.

9. __X___ The prior application is assigned to Motorola, Inc.

10. __X___ The power of attorney in the prior application is to

____

____

a. __X__ The power appears in the original papers of the prior application.

b. _____ Since the power does not appear in the original papers, a copy of the power in the prior application is enclosed.

c. _____ Recognize as associate attorney and address all future communication to:
James A. Lamb

11. _____ Priority under 35 USC 119 is claimed on the basis of prior _______ Application No. _______, filed __________.

12. _____ A new oath/declaration is enclosed for a continuation-in-part application.

13. __X___ An Information Disclosure Statement and PTO form 1449 are enclosed.

Kelly A. Gardner
Kelly A. Gardner
Attorney for Applicants
Registration No. 35,147
Phone: (561) 739-2862

Please direct all correspondence to:

Motorola, Inc., IP Law Department MS 96, 1500 Gateway Blvd., Boynton Beach, FL 33426-8753

A

08/764656

#7
ME
3-18-97

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re application of: : Date: December 3, 1996

LEITCH, et al. :

Docket No.: PT00600UC01 :

Filed: Concurrently Herewith :

For: VOICE COMPRESSION METHOD AND APPARATUS IN A COMMUNICATION SYSTEM

INFORMATION DISCLOSURE STATEMENT
PURSUANT TO 37 C.F.R. §§ 1.56, 1.97, and 1.98

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

Sir:

Applicant submits herewith the art listed below of which the Applicant is aware, which the Applicant believes may be material to the examination of the subject application and in respect of which there may be a duty to disclose in accordance with 37 C.F.R. § 1.56. This citation of information, also appearing on the attached Form PTO-1449, "List of Art Cited by Applicant" is made pursuant to 37 C.F.R. §§ 1.56, 1.97, and 1.98.

A copy of the art listed below is enclosed herewith, unless the art is cumulative as noted below or the art has been cited in a prior application from which an earlier filing date is claimed for the subject application, the earlier application noted below.

The filing of this Information Disclosure Statement shall not be construed as a representation that a search has been made, an admission that the information cited is, or is considered to be, material to patentability, or that no other material information exists. Further, the filing of this Information Disclosure Statement shall not be construed as an admission against interest in any manner.

Pursuant to 37 C.F.R. § 1.98, as amended March 16, 1992, no explanation of the relevance of the English language references is presented.

PATENT APPLICATION
PT00600UC01

Patents

U.S. Patent No. 5,121,391 issued June 9, 1992 to Paneth, et al.
U.S. Patent No. 4,955,083 issued September 4, 1990 to Phillips, et al.
U.S. Patent No. 4,134,069 issued January 9, 1979 to Shiki.

The attorney signing below is making this Information Disclosure Statement on the basis of information supplied by the inventor, an individual associated with the filing and prosecution of the subject application and/or information in the attorney's files. The citation of this information does not constitute either an admission of priority or a waiver of any right applicant may have under applicable statutes, Rules of Practice in patent cases, or otherwise.

Respectfully submitted,

LEITCH, et al.

By: Kelly Gardner

Kelly A. Gardner
Attorney for Applicant
Registration No. 35,147
Tel. (561) 739-2862
Fax. (561) 739-2825

MOTOROLA, INC.
Patent Department, MS96
1500 Gateway Boulevard
Boynton Beach, FL 33426-8292

F/ RECEIVED

MAR 2 8 1997

GROUP 2400

OFFICIAL

FORM PTO-1449 U.S. DEPARTMENT OF COMMERCE, PATENT AND TRADEMARK OFFICE

LIST OF ART CITED BY APPLICANT

(Use several sheets if necessary)

ATTY. DOCKET NO.	SERIAL NO.	FILING DATE
PT00600UC01	08/764,656	12/11/96
APPLICANT		**GROUP**
LEITCH, et al.		2412

U.S. PATENT DOCUMENTS

EXAMINER INITIAL		DOCUMENT NUMBER	DATE	NAME	CLASS	SUBCLASS	FILING DATE IF APPROPRIATE
DS	AA	5,121,391	6/9/92	Paneth, et al.	—	—	
DS	AB	4,955,083	9/4/90	Phillips, et al.	—	—	
DS	AC	4,134,069	1/9/79	Shiki	—	—	
	AD						
	AE						
	AF						
	AG						
	AH						
	AI						
	AJ						
	AK						

FOREIGN PATENT DOCUMENTS

		DOCUMENT NUMBER	DATE	COUNTRY	CLASS	SUBCLASS	TRANSLATION YES	TRANSLATION NO
	AL							
	AM							
	AN							
	AO							

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

	AP	
	AR	
	AS	
	AT	

EXAMINER	DATE CONSIDERED
Dennita Smith	4/8/97

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

SENT BY:MOTOROLA PAGING ; 3-28-97 ; 1:12PM ; BOYNTON→ 17033059731;# 2/ 2

FORM PTO-1449 U.S. DEPARTMENT OF COMMERCE, PATENT AND TRADEMARK OFFICE

LIST OF ART CITED BY APPLICANT

(Use several sheets if necessary)

ATTY. DOCKET NO.	SERIAL NO.	FILING DATE
PT00600UC01	08/764,656	12/11/96
APPLICANT		GROUP
LEITCH, et al.		2412

U.S. PATENT DOCUMENTS

EXAMINER INITIAL		DOCUMENT NUMBER	DATE	NAME	CLASS	SUBCLASS	FILING DATE IF APPROPRIATE
	AA	5,121,391	6/9/92	Paneth, et al.			
	AB	4,955,083	9/4/90	Phillips, et al.			
	AC	4,134,069	1/9/79	Shiki			
	AD						
	AE						
	AF						
	AG						
	AH						
	AI						
	AJ						
	AK						

Duplicate

FOREIGN PATENT DOCUMENTS

		DOCUMENT NUMBER	DATE	COUNTRY	CLASS	SUBCLASS	TRANSLATION YES	TRANSLATION NO.
	AL							
	AM							
	AN							
	AO							

OTHER PRIOR ART (*Including Author, Title, Date, Pertinent Pages, Etc.*)

	AP	
	AR	
	AS	
	AT	

EXAMINER	DATE CONSIDERED

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

UNITED STATES PATENT & TRADEMARK OFFICE
Washington, D.C. 20231

REQUEST FOR PATENT FEE REFUND

Date of Request: 1-13-97 | 2 Serial/Patent # 08/764656

3 Please refund the following fee(s):		4 PAPER NUMBER	5 DATE FILED	6 AMOUNT
✓	Filing		12-10-96	$ 22
	Amendment			$
	Extension of Time			$
	Notice of Appeal/Appeal			$
	Petition			$
	Issue			$
	Cert of Correction/Terminal Disc.			$
	Maintenance			$
	Assignment			$
	Other			$
		7 TOTAL AMOUNT OF REFUND		$ 22

8 TO BE REFUNDED BY:

	Treasury Check
	Credit Deposit A/C #:
9	13--4778

10 REASON:

✓	Overpayment
	Duplicate Payment
	No Fee Due (Explanation):

11 REFUND REQUESTED BY:

TYPED/PRINTED NAME: J.M. Tsay TITLE: LIE

SIGNATURE: [signature] PHONE: 308-1202

OFFICE: OIPE T-2

****

THIS SPACE RESERVED FOR FINANCE USE ONLY:

APPROVED: [signature] DATE: 1/16/97

Instructions for completion of this form appear on the back. After completion, attach white and yellow copies to the official file and mail or hand-carry to:

Office of Finance
Refund Branch
Crystal Park One, Room 802B

FORM PTO 1577
(01/90)

UNITED STATES PATENT & TRADEMARK OFFICE
Washington, D.C. 20231

REQUEST FOR PATENT FEE REFUND

1 Date of Request: 1-13-97 | 2 Serial/Patent # 08764656

3 Please refund the following fee(s):		4 PAPER NUMBER	5 DATE FILED	6 AMOUNT
✓	Filing		12-10-96	$ 22
	Amendment			$
	Extension of Time			$
	Notice of Appeal/Appeal			$
	Petition			$
	Issue			$
	Cert of Correction/Terminal Disc.			$
	Maintenance			$
	Assignment			$
	Other			$

7 TOTAL AMOUNT OF REFUND $ 22

8 TO BE REFUNDED BY:

	Treasury Check
	Credit Deposit A/C #:
9	13-4778

10 REASON:

✓	Overpayment
	Duplicate Payment
	No Fee Due (Explanation):

11 REFUND REQUESTED BY:

TYPED/PRINTED NAME: T. Tsy TITLE: RJ E.

SIGNATURE: [signature] PHONE: 308-1268

OFFICE: OIPE T-2

THIS SPACE RESERVED FOR FINANCE USE ONLY:

APPROVED: [signature] DATE: 1/16/97

Instructions for completion of this form appear on the back. After completion, attach white and yellow copies to the official file and mail or hand-carry to:

Office of Finance
Refund Branch
Crystal Park One, Room 802B

FORM PTO 1577 (01/90)

RT18104 01/29/97 08764656 18 180 101 22.00CR

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

SERIAL NUMBER	FILING DATE	FIRST NAMED APPLICANT		ATTORNEY DOCKET NO.
08/764,656	12/11/96	LEITCH	C	PT00600UC01

B3M1/0319

MOTOROLA INC
IP LAW DEPARTMENT MS 96
1500 GATEWAY BLVD
BOYNTON BEACH FL 33426-8753

EXAMINER	
SMITH,D	
ART UNIT	PAPER NUMBER
2414	#8

DATE MAILED: 03/19/97

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

Commissioner of Patents

See Attachments

ELLIS B. RAMIREZ
PRIMARY EXAMINER
GROUP 2400

PTOL-90 (Rev. 6/84)

1 - PATIENT APPLICATION FILE COPY

Notice of Allowability	Application No. 08/764,656	Applicant(s) Leitch et al
	Examiner Demetra R. Smith	Group Art Unit 2414

All claims being allowable, PROSECUTION ON THE MERITS IS (OR REMAINS) CLOSED in this application. If not included herewith (or previously mailed), a Notice of Allowance and Issue Fee Due or other appropriate communication will be mailed in due course.

☒ This communication is responsive to *12/11/96*.

☒ The allowed claim(s) is/are *1-24*.

☐ The drawings filed on ________ are acceptable.

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

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

☐ received.

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

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

*Certified copies not received: ________.

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

A SHORTENED STATUTORY PERIOD FOR RESPONSE to comply with the requirements noted below is set to EXPIRE **THREE MONTHS** FROM THE "DATE MAILED" of this Office action. Failure to timely comply will result in ABANDONMENT of this application. Extensions of time may be obtained under the provisions of 37 CFR 1.136(a).

☐ Note the attached EXAMINER'S AMENDMENT or NOTICE OF INFORMAL APPLICATION, PTO-152, which discloses that the oath or declaration is deficient. A SUBSTITUTE OATH OR DECLARATION IS REQUIRED.

☐ Applicant MUST submit NEW FORMAL DRAWINGS

☐ because the originally filed drawings were declared by applicant to be informal.

☐ including changes required by the Notice of Draftsperson's Patent Drawing Review, PTO-948, attached hereto or to Paper No. ____.

☐ including changes required by the proposed drawing correction filed on ________, which has been approved by the examiner.

☐ including changes required by the attached Examiner's Amendment/Comment.

Identifying indicia such as the application number (see 37 CFR 1.84(c)) should be written on the reverse side of the drawings. The drawings should be filed as a separate paper with a transmittal lettter addressed to the Official Draftsperson.

☐ Note the attached Examiner's comment regarding REQUIREMENT FOR THE DEPOSIT OF BIOLOGICAL MATERIAL.

Any response to this letter should include, in the upper right hand corner, the APPLICATION NUMBER (SERIES CODE/SERIAL NUMBER). If applicant has received a Notice of Allowance and Issue Fee Due, the ISSUE BATCH NUMBER and DATE of the NOTICE OF ALLOWANCE should also be included.

Attachment(s)

☒ Notice of References Cited, PTO-892

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

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

☐ Notice of Informal Patent Application, PTO-152

☐ Interview Summary, PTO-413

☐ Examiner's Amendment/Comment

☐ Examiner's Comment Regarding Requirement for Deposit of Biological Material

☒ Examiner's Statement of Reasons for Allowance

DETAILED ACTION

Allowable Subject Matter

1. Claims 1-24 are allowed.

2. The following is an examiner's statement of reasons for allowance: The prior art does not show separately or in combination the limitations of subchanneling the voice communication resource and simultaneously placing at least one of each of the plurality of voice signals on a subchannel; compressing the time of each of the voice signals within each of the subchannels; wherein the step of compressing the time of each of the voice signals further comprises the step of using Waveform Similarity based Overlap-Add (WSOLA) time compression on the voice signals; compressing the bandwidth of plurality of voice signals by subchanneling the voice communication resource and modulating each of the plurality of voice signals about a plurality of pilot signals within the voice communication resource; a processing device for demodulating the received processed signal using single side band demodulation and time-scale expansion to provide a reconstructed signal; subchannelizing the communication resource into a predetermined number of subchannels; and a digital signal processor for performing single sideband demodulation and at least one of the functions of filtering a pilot carrier, performing automatic gain control using a feedforward loop, or decompanding of the digitized received signal to provide a processed signal.

Any comments considered necessary by applicant must be submitted no later than the payment of the issue fee and, to avoid processing delays, should preferably accompany the issue fee. Such submissions should be clearly labeled "Comments on Statement of Reasons for Allowance."

Conclusion

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

Dejmek et al. ('898) discloses a method for time compressing a voice message in a voice messaging method for selective call receivers.

Hieatt, III ('215) discloses s selective call transceiver unit, selective call base station in which the receiver processes the radio signals and produces output demodulated information. The demodulated information is coupled into the input of a microprocessor which processed the information and incorporates conventional analog-to-digital and digital-to-voice message from the receiver.

Liberti, Jr. et al. discloses a selective call base station for use in a selective call communication system, comprising a transmitter coupled to the controller for transmitting selective call signals to selective call receivers operating in the selective call communication system, a receiver coupled to the controller for receiving a modulated radio signal from a transmitter within one of the selective call receivers, an analog-to-digital converter for digitizing the modulated radio signal and a symbol demodulator coupled to the carrier frequency estimator for demodulating a symbol value corresponding to one of the serially transmitted symbol intervals.

Orlen et al. ('981) discloses a selective call communication system that includes an encoder for encoding and compressing the data to form compressed data, and a controller for transferring the compressed data to a selective call terminal.

Puckette ('589) discloses a digitally implemented single sideband (SSB) transmitter simultaneously generates a desired signaling waveform in response to a single data signal input, multiplies it by a carrier function, generates the Hilbert transform of the signaling waveform and multiplies it by a quadrature carrier function, and sums the two products to produce the SSB output.

Wong ('174) discloses an analog signal stacking method and apparatus permits transmission of n analog signals of given spectrum width over a narrow band communication circuit having bandwidth approximately equal to the spectrum width of one analog signal by compressing the spectrum of each signal by a factor of 1/n.

4. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Demetra R. Smith whose telephone number is (703) 308-6989. The examiner can normally be reached on Monday-Friday from 8:00 to 5:00.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, E. Todd Voeltz, can be reached on (703) 305-9714. The fax phone number for this Group is (703) 305-9731.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the Group receptionist whose telephone number is (703) 305-3800.

Demetra R Smith

Demetra R. Smith

March 12, 1997

Notice of References Cited	Application No. 08/764,656	Applicant(s) Leitch et al
	Examiner Demetra R. Smith	Group Art Unit 2414	Page 1 of 1

U.S. PATENT DOCUMENTS

*		DOCUMENT NO.	DATE	NAME	CLASS	SUBCLASS
x	A	3,793,589	2-19-74	Puckette	325	137
x	B	4,586,174	4-29-86	Wong	370	69.1
x	C	5,533,062	7-2-96	Liberti, Jr. et al	375	334
x	D	5,068,898	11-26-91	Dejmek et al	381	29
x	E	5,387,981	2-7-96	Orlen et al	358	400
x	F	5,535,215	7-9-96	Hieatt, III	370	95.1
	G					
	H					
	I					
	J					
	K					
	L					
	M					

FOREIGN PATENT DOCUMENTS

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

NON-PATENT DOCUMENTS

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

Demetra Smith 3/12/97

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

U. S. Patent and Trademark Office
PTO-892 (Rev. 9-95)

Notice of References Cited

Part of Paper No. 8

BEST COPY

UNITED STATES DEPARTMENT OF COMMERCE
Patent and Trademark Office

Address: Box ISSUE FEE
ASSISTANT COMMISSIONER FOR PATENTS
WASHINGTON, D.C. 20231

NOTICE OF ALLOWANCE AND ISSUE FEE DUE

B3M1/0319

MOTOROLA INC
IP LAW DEPARTMENT MS 96
1500 GATEWAY BLVD
BOYNTON BEACH FL 33426-8753

APPLICATION NO.	FILING DATE	TOTAL CLAIMS	EXAMINER AND GROUP ART UNIT		DATE MAILED
08/764,656	12/11/96	024	SMITH, D	2414	03/19/97

First Named Applicant: LEITCH, CLIFFORD D.

TITLE OF INVENTION: VOICE COMPRESSION METHOD AND APPARATUS IN A COMMUNICATION SYSTEM

ATTY'S DOCKET NO.	CLASS-SUBCLASS	BATCH NO.	APPLN. TYPE	SMALL ENTITY	FEE DUE	DATE DUE
PT00600UC01	364-514.00R	K90	UTILITY	NO	$1290.00	06/19/97

THE APPLICATION IDENTIFIED ABOVE HAS BEEN EXAMINED AND IS ALLOWED FOR ISSUANCE AS A PATENT. PROSECUTION ON THE MERITS IS CLOSED.

THE ISSUE FEE MUST BE PAID WITHIN THREE MONTHS FROM THE MAILING DATE OF THIS NOTICE OR THIS APPLICATION SHALL BE REGARDED AS ABANDONED. THIS STATUTORY PERIOD CANNOT BE EXTENDED.

HOW TO RESPOND TO THIS NOTICE:

I. Review the SMALL ENTITY status shown above. If the SMALL ENTITY is shown as yes, verify your current SMALL ENTITY status:

A. If the status is changed, pay twice the amount of the FEE DUE shown and notify the Patent and Trademark Office of the change in status, or

B. If the status is the same, pay the FEE DUE shown above.

If the SMALL ENTITY is shown as NO:

A. Pay FEE DUE shown above, or

B. File verified statementof Small Entity Status before, or with, payment of 1/2 the FEE DUE shown above.

II. Part B of this notice should be completed and returned to the Patent and Trademark Office (PTO) with your ISSUE FEE. Even if the ISSUE FEE has already been paid by charge to deposit account, Part B should be completed and returned. If you are charging the ISSUE FEE to your deposit account, section "6b" of Part B should be completed.

III. All communications regarding this application must give application number and batch number. Please direct all communication prior to issuance to Box ISSUE FEE unless advised to the contrary.

IMPORTANT REMINDER: ***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.***

3. PATENT AND TRADEMARK OFFICE COPY

PTOL-85 (REV. 05-96)(0651-0033)

*U.S. GPO: 1996-404-496/40511

11
M.E
5-1597

UNITED STATES PATENT AND TRADEMARK OFFICE PATENT APPLICATION

APPLICANT: Leitch, et al. EXAMINER: D. Smith

SERIAL NO.: 08/764,656 ART UNIT: 2414

FILED: 12/11/96 CASE NO.: PT00600UC01

APPLICATION: VOICE COMPRESSION METHOD AND APPARATUS IN A COMMUNICAITON SYSTEM

COMMUNICATION TO EXAMINER

Commissioner of Patents and Trademarks
Washington, DC 20231

Sir:

Attached is a confirmation copy of a facsimile transmitted on Mar. 28, 1997.

Respectfully submitted,
Leitch, et al.

By: James A. Lamb

James A. Lamb
Agent for Applicants
Registration No. 38,529
Phone: (561) 739-2860
Fax: (561) 739-2825

Please direct all correspondence to:

Motorola Inc.
Intellectual Property Department MS 96
1500 Gateway Blvd.
Boynton Beach, FL 33426-8292

FAX RECEIVED

MAR 2 8 1997

GROUP 2400

#9 Turner B 4/23/97

INTELLECTUAL PROPERTY DEPARTMENT

MOTOROLA INC.

PAGING AND TELEPOINT SYSTEMS GROUP
BOYNTON BEACH, FLORIDA 33426

FAX	MESSAGE TRANSMITTAL FAX: 561-739-2825 VERIFY 561-739-2860

To:	Examiner Demetra Smith	From:	James A. Lamb
Phone:	(703) 308-6989	Agent:	38, 529
FAX:	(703)-305-9731	Phone:	407-739-2721

Application number: 08/764,656 - Leich et al.; filed 12/11/96

Number of Pages: 2

March 28, 1997

Examiner Smith,

Transmitted herewith is a form 1449, listing the prior art included with the continuation filing of our case number PT00600UC01 made on Dec. 11, 1996 and itemized in the Information Disclosure Statement included therewith. Per our telephone conversation of May 26, 1994, please provide an acknowledged copy of the form 1449 and amend your Notice of Allowance to include mention of the art, or provide equivalent documentation. Should you have any questions, call between 9:00 and 5:00, Monday through Friday, at (561) 739-2862. I will follow this up with a mailed confirmation copy.

Sincerely,

James A. Lamb

NOTICE: This facsimile transmission may contain information that is confidential, privileged or exempt from disclosure under applicable law. It is intended only for the person to whom it is addressed. Unauthorized use, disclosure, copying or distribution may expose you to legal liability. If you have received this transmission in error, please immediately notify us by telephone (collect) to arrange for return of the documents received and any copies made. Thank you.

INTELLECTUAL PROPERTY DEPARTMENT

MOTOROLA INC.

PAGING AND TELEPOINT SYSTEMS GROUP
BOYNTON BEACH, FLORIDA 33426

FAX	MESSAGE TRANSMITTAL FAX: 561-739-2825 VERIFY 561-739-2860

To:	Examiner Demetra Smith	From:	James A. Lamb
Phone:	(703) 308-6989	Agent:	38,529
FAX:	(703)-305-9731	Phone:	407-739-2721

Application number: 08/764,656 - Leich et al.; filed 12/11/96

Number of Pages: 2

March 28, 1997

Examiner Smith,

Transmitted herewith is a form 1449, listing the prior art included with the continuation filing of our case number PT00600UC01 made on Dec. 11, 1996 and itemized in the Information Disclosure Statement included therewith. Per our telephone conversation of May 26, 1994, please provide an acknowledged copy of the form 1449 and amend your Notice of Allowance to include mention of the art, or provide equivalent documentation. Should you have any questions, call between 9:00 and 5:00, Monday through Friday, at (561) 739-2862. I will follow this up with a mailed confirmation copy.

Sincerely,

James A. Lamb

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

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.
08/764,656	12/11/96	LEITCH	L PT00600UC01

B3M1/0424

MOTOROLA INC
IP LAW DEPARTMENT MS 96
1500 GATEWAY BLVD
BOYNTON BEACH FL 33426-8753

EXAMINER	
SMITH, D	
ART UNIT	**PAPER NUMBER**
2414	6

DATE MAILED: 04/24/97

Please find below and/or attached an Office communication concerning this application or proceeding.

Commissioner of Patents and Trademarks

See Attachments

EMANUEL T. VOELTZ
SUPERVISORY PATENT EXAMINER
GROUP 2400

PTO-90C (Rev. 2/95) ☆U.S.GOVERNMENT PRINTING OFFICE 1996-411-516/40275 1- File Copy

Notice of Allowability	Application No. 08/764,656	Applicant(s) Leitch et al
	Examiner Dematra R. Smith	Group Art Unit 2414

All claims being allowable, PROSECUTION ON THE MERITS IS (OR REMAINS) CLOSED in this application. If not included herewith (or previously mailed), a Notice of Allowance and Issue Fee Due or other appropriate communication will be mailed in due course.

☒ This communication is responsive to *12/11/96*

☒ The allowed claim(s) is/are *1-24*

☐ The drawings filed on ________________ are acceptable.

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

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

☐ received.

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

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

*Certified copies not received: ________________

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

A SHORTENED STATUTORY PERIOD FOR RESPONSE to comply with the requirements noted below is set to EXPIRE **THREE MONTHS** FROM THE "DATE MAILED" of this Office action. Failure to timely comply will result in ABANDONMENT of this application. Extensions of time may be obtained under the provisions of 37 CFR 1.136(a).

☐ Note the attached EXAMINER'S AMENDMENT or NOTICE OF INFORMAL APPLICATION, PTO-152, which discloses that the oath or declaration is deficient. A SUBSTITUTE OATH OR DECLARATION IS REQUIRED.

☐ Applicant MUST submit NEW FORMAL DRAWINGS

☐ because the originally filed drawings were declared by applicant to be informal.

☐ including changes required by the Notice of Draftsperson's Patent Drawing Review, PTO-948, attached hereto or to Paper No. ______.

☐ including changes required by the proposed drawing correction filed on ________________, which has been approved by the examiner.

☐ including changes required by the attached Examiner's Amendment/Comment.

Identifying indicia such as the application number (see 37 CFR 1.84(c)) should be written on the reverse side of the drawings. The drawings should be filed as a separate paper with a transmittal lettter addressed to the Official Draftsperson.

☐ Note the attached Examiner's comment regarding REQUIREMENT FOR THE DEPOSIT OF BIOLOGICAL MATERIAL.

Any response to this letter should include, in the upper right hand corner, the APPLICATION NUMBER (SERIES CODE/SERIAL NUMBER). If applicant has received a Notice of Allowance and Issue Fee Due, the ISSUE BATCH NUMBER and DATE of the NOTICE OF ALLOWANCE should also be included.

Attachment(s)

☒ Notice of References Cited, PTO-892

☒ Information Disclosure Statement(s), PTO-1449, Paper No(s). 7

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

☐ Notice of Informal Patent Application, PTO-152

☐ Interview Summary, PTO-413

☐ Examiner's Amendment/Comment

☐ Examiner's Comment Regarding Requirement for Deposit of Biological Material

☒ Examiner's Statement of Reasons for Allowance

DETAILED ACTION

Allowable Subject Matter

1. Claims 1-24 are allowed.

2. The following is an examiner's statement of reasons for allowance: The prior art does not show separately or in combination the limitations of subchanneling the voice communication resource and simultaneously placing at least one of each of the plurality of voice signals on a subchannel; compressing the time of each of the voice signals within each of the subchannels; wherein the step of compressing the time of each of the voice signals further comprises the step of using Waveform Similarity based Overlap-Add (WSOLA) time compression on the voice signals; compressing the bandwidth of plurality of voice signals by subchanneling the voice communication resource and modulating each of the plurality of voice signals about a plurality of pilot signals within the voice communication resource; a processing device for demodulating the received processed signal using single side band demodulation and time-scale expansion to provide a reconstructed signal; subchannelizing the communication resource into a predetermined number of subchannels; and a digital signal processor for performing single sideband demodulation and at least one of the functions of filtering a pilot carrier, performing automatic gain control using a feedforward loop, or decompanding of the digitized received signal to provide a processed signal.

Any comments considered necessary by applicant must be submitted no later than the payment of the issue fee and, to avoid processing delays, should preferably accompany the issue fee. Such submissions should be clearly labeled "Comments on Statement of Reasons for Allowance."

Conclusion

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

Dejmek et al. ('898) discloses a method for time compressing a voice message in a voice messaging method for selective call receivers.

Hieatt, III ('215) discloses s selective call transceiver unit, selective call base station in which the receiver processes the radio signals and produces output demodulated information. The demodulated information is coupled into the input of a microprocessor which processed the information and incorporates conventional analog-to-digital and digital-to-voice message from the receiver.

Liberti, Jr. et al. discloses a selective call base station for use in a selective call communication system, comprising a transmitter coupled to the controller for transmitting selective call signals to selective call receivers operating in the selective call communication system, a receiver coupled to the controller for receiving a modulated radio signal from a transmitter within one of the selective call receivers, an analog-to-digital converter for digitizing the modulated radio signal and a symbol demodulator coupled to the carrier frequency estimator for demodulating a symbol value corresponding to one of the serially transmitted symbol intervals.

Orlen et al. ('981) discloses a selective call communication system that includes an encoder for encoding and compressing the data to form compressed data, and a controller for transferring the compressed data to a selective call terminal.

Puckette ('589) discloses a digitally implemented single sideband (SSB) transmitter simultaneously generates a desired signaling waveform in response to a single data signal input, multiplies it by a carrier function, generates the Hilbert transform of the signaling waveform and multiplies it by a quadrature carrier function, and sums the two products to produce the SSB output.

Wong ('174) discloses an analog signal stacking method and apparatus permits transmission of n analog signals of given spectrum width over a narrow band communication circuit having bandwidth approximately equal to the spectrum width of one analog signal by compressing the spectrum of each signal by a factor of 1/n.

4. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Demetra R. Smith whose telephone number is (703) 308-6989. The examiner can normally be reached on Monday-Friday from 8:00 to 5:00.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, E. Todd Voeltz, can be reached on (703) 305-9714. The fax phone number for this Group is (703) 305-9731.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the Group receptionist whose telephone number is (703) 305-3800.

Demetra R. Smith

April 8, 1997

EMANUEL T. VOELTZ
SUPERVISORY PATENT EXAMINER
GROUP 2400

Notice of References Cited	Application No. 08/764,656	Applicant(s) Leitch et al
	Examiner Demetra R. Smith	Group Art Unit 2414 — Page 1 of 1

U.S. PATENT DOCUMENTS

*		DOCUMENT NO.	DATE	NAME	CLASS	SUBCLASS
x	A	3,793,589	2-19-74	Puckette	325	137
x	B	4,586,174	4-29-86	Wong	370	69.1
x	C	5,533,062	7-2-96	Liberti, Jr. et al	375	334
x	D	5,068,898	11-26-91	Dejmek et al	381	29
x	E	5,387,981	2-7-96	Orlen et al	358	400
x	F	5,535,215	7-9-96	Hieatt, III	370	95.1
	G					
	H					
	I					
	J					
	K					
	L					
	M					

FOREIGN PATENT DOCUMENTS

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

NON-PATENT DOCUMENTS

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

Demetra Smith 4/8/97

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

U. S. Patent and Trademark Office
PTO-892 (Rev. 9-95) Notice of References Cited Part of Paper No. 2

RECEIVED MAY 22 1997

OFFICIAL

RECEIVED MAY 22 1997 GROUP 2300

INTELLECTUAL PROPERTY DEPARTMENT

MOTOROLA INC.

PAGING AND TELEPOINT SYSTEMS GROUP
BOYNTON BEACH, FLORIDA 33426

FAX

MESSAGE TRANSMITTAL

FAX: 561-739-2825
VERIFY 561-739-2860

To:	Mr. E. Todd Voeltz FOR Examiner Demetra Smith	From:	James A. Lamb
Phone:	(703) 308-6989	Agent:	38, 529
FAX:	(703)-305-9731	Phone:	407-739-2862

Application number: 08/764,656

Number of Pages: 9

May 22, 1997

Examiner Smith,

In accordance with our telephone conversation this morning, I am transmitting herewith a set of claims corresponding to those used to amend the PCT case, PCT/US96/)0095.

Amended claims 1, 2-12, 13, and 14-20 correspond respectively, to claims 1, 4-14, 16, and 18-24 of the orginally filed claims. Claims 2-3, 15, and 17 as originally filed were cancelled. In the amended claims, independent claims 1, 7, 10, 13, and 19 are amended. In the amended claims, independent claims 14 and 20, and dependent claims 2-6, 8-9, 11-12, and 15-18 are unchanged from the claims as filed.

<u>Amended claim 1 differs</u> from the amended claim 1 as supplied in the PCT amendment for reasons which follow:

OFFICIAL

The PCT amended claim 1 read as follows:

A method for compressing a plurality of voice signals within a voice communication resource having a given bandwidth within a voice communication system, comprising the steps of:

(a) subchanneling the voice communication resource and simultaneously placing placing pairs of the plurality of voice signals on a subchannel using single sideband modulation;

(b) modulating each of the pairs of the plurality of voice signals about one of a plurality of pilot signals within a subchannel within the voice communication resource; and

(c) compressing the time of each of the voice signals within each of the subchannels, wherein the result of steps (a), (b), and (c) provides a compressed voice signal.

Step (a) of the PCT amended claim could infer placing more than one pair of voice signals on a single subchannel, which is not described in the detailed description. This is clarified in the amended claims supplied below by changing claim 1 to include a description of placing a pair of the plurality of voice channels on a subchannel in step (a) to avoid this inference. Other changes are made in claim 1 to clarify antecedent bases.

Amended Claim 7 as supplied herewith is modified in the manner of amended claim 1 supplied herewith.

All other claims are exactly as supplied in the PCT amendment.

The abstract was not modified in the PCT amendment. I have included an Abstract modified to correspond to claim 1.

No new matter was added. If you require the claims in a different form or need any additional information, please do not hestitate to call.

Sincerely,
James A. Lamb
James A. Lamb

NOTICE: This facsimile transmission may contain information that is confidential, privileged or exempt from disclosure under applicable law. It is intended only for the person to whom it is addressed. Unauthorized use, disclosure, copying or distribution may expose you to legal liability. If you have received this transmission in error, please immediately notify us by telephone (collect) to arrange for return of the documents received and any copies made. Thank you.

In the claims:

Please replace the claims as filed with the following claims:

1. (Amended) A method for compressing a plurality of voice signals within a voice communication resource having a given bandwidth within a voice communication system, comprising the steps of:

(a) subchanneling the voice communication resource into a plurality of subchannels and simultaneously placing a pair of the plurality of voice signals on a subchannel using single sideband modulation;

(b) modulating the pair of the plurality of voice signals about a pilot signal within the subchannel within the voice communication resource; and

(c) compressing the time of each of the voice signals within the plurality of subchannels, wherein the result of steps (a), (b), and (c) provides a compressed voice signal.

2. The method of claim 1, wherein the step of subchanneling further comprises the step of using quadrature amplitude modulation.

3. The method of claim 1, wherein the step of compressing the time of each of the voice signals further comprises the step using time-scale compression on the voice signals.

4. The method of claim 1, wherein the step of compressing the time of each of the voice signals further comprises the step of using Waveform Similarity based Overlap-Add (WSOLA) time compression on the voice signals.

5. The method of claim 1, wherein the step of compressing the time of each of the voice signals further comprises the speaker dependent steps of identifying pitch periods within each of the voice signals and transmitting data from one pitch period to alter a time-scaling factor.

6. The method of claim 1, wherein the step of compressing the time of each of the voice signals comprises the step of using a speaker dependent modification of the Waveform Similarity based Overlap-Add (WSOLA) time compression technique on the voice signals.

7. (Amended) A method for compressing a plurality of voice signals within a voice communication resource within a voice communication system, comprising the steps of:

(a) subchanneling the voice communication resource into a plurality of subchannels and simultaneously placing a pair of the plurality of voice signals on a subchannel using single sideband modulation;

(b) modulating the pair of the plurality of voice signals about a pilot signal within the subchannel within the voice communication resource; and

(c) compressing the time of each of the voice signals within the plurality of subchannels, wherein the result of steps (a), (b), and (c) provides a compressed voice signal for transmission via a transmitter.

8. The method of claim 7, wherein the method further comprises the step at the transmitter of transmitting the compressed voice signal to a plurality of selective call receivers.

9. The method of claim 7, wherein the method further comprises the step of receiving the compressed voice signal and demodulating the compressed bandwidth signals at one of the the plurality of selective call receivers.

SENT BY:MOTOROLA PAGING ; 5-22-97 ; 7:57PM ; BOYNTON→ 17033059731;# 4/ 9

10. (Amended) A communication system using voice compression having at least one transmitter base station and a plurality of selective call receivers, comprising:

at the transmitter base station:

an input device for receiving an audio signal;

a processing device for compressing the audio signal using time-scale compression and a single side band modulation technique to provide a processed signal;

a pilot carrier signal generator that generates a pilot carrier for a pair of single side band signals which includes the processed signal, wherein the pilot carrier serves as an amplitude and phase reference for distortion that occurs as a result of channel aberrations; and

a transmitter for transmitting the processed signal;

at each of the plurality of selective call receivers:

a selective call receiver for receiving the transmitted processed signal;

a receiver circuit for detecting, filtering and responding to the amplitude and phase reference generated by the pilot carrier signal generator;

a processing device for demodulating the received processed signal using single side band demodulation and time-scale expansion to provide a reconstructed signal; and

an amplifier for amplifying the reconstructed signal into an reconstructed audio signal.

11. The communication system of claim 10, wherein the single sideband modulation technique provides for the transmission of a single message split between an upper sideband and a lower sideband.

12. The communication system of claim 10, wherein the single sideband modulation technique provides for the transmission of a single message repeated on an upper sideband and lower sideband.

13. (Amended) A selective call receiver for receiving compressed voice signals, comprising

a selective call receiver for receiving a transmitted processed signal that includes compressed voice signals that have been compressed using time-scale compression;

a processing device for demodulating the received processed signal, wherein said processing device demodulates both an upper and a lower sideband of a subchannel, the upper and lower sidebands having independent information therein, and wherein said processing device uses time-scale expansion to provide a reconstructed signal;

a receiver circuit for detecting, filtering and responding to the amplitude and phase reference generated by a pilot carrier signal generator in a transmitter at a base station; and

an amplifier for amplifying the reconstructed signal into an reconstructed audio signal.

14. A selective call paging base station for transmitting selective call signals on a communication resource having a predetermined bandwidth, comprising:

an input device for receiving a plurality of audio signals;

a means for subchannelizing the communication resource into a predetermined number of subchannels;

an amplitude compression and filtering module for each subchannel for compressing the amplitude of the respective audio signal and filtering the respective audio signal;

a time compression module for compressing the time of the respective audio signal for each subchannel; and

a quadrature amplitude modulation transmitter for transmitting the processed signal.

15. The selective call paging base station of claim 14, wherein the input device for receiving a plurality of audio signals comprises a paging terminal for receiving phone messages or data messages from a computing device.

16. The selective call paging base station of claim 14, wherein the amplitude compression and filtering module comprises an anti-alias filter coupled to an analog-to-digital converter coupled to a band-pass filter coupled to an automatic gain controller and clipper circuit.

17. The selective call paging base station of claim 14, wherein the time compression module comprises a processing device for compressing the audio signal using a time-scale compression technique.

18. The selective call paging base station of claim 14, wherein the time compression module comprises a processing device for compressing the audio signal using a WSOLA time compression technique.

19. (Amend:d) A selective call receiver unit for receiving compressed voice selective call signals, comprising:

a recei er having a analog to digital converter for providing a digitized received signal;

a digit l signal processor for performing single sideband demodulation of a subchannel having a pilot carrier and independent information on an upper and a lower sideb nd of a subchannel, wherein the digital signal processor also performs the functions of filtering the pilot carrier, performing automatic gain control using a feedforwarc loop, and decompanding the digitized received signal to provide a processed si gnal; and

a digit l to analog converter and reconstruction filter for converting the processed signal into digitized audio signal; and

an amp lifier for amplifying the digitized audio signal.

20. A com nunication base station, comprising:

a term nal for receiving an audio speech signal;

an ana og to digital converter for converting the audio speech signal into a digitized sp ech signal;

a digit l signal processor for processing the digitized speech signal by performing the function of splitting the digitized speech signal and at least one of the functions of bandpass filtering, automatic gain control , time scaling, companding, or buffering; and

a trans nitter having at least a Hilbert transform filter coupled to a digital to analog converter co upled to a reconstruction filter coupled to a quadrature amplitude modulator which is coupled to a radio frequency power amplifier. —

In the Abstract

Please cancel the Abstract as filed and use the following abstract:

--The present invention comprises a method for compressing a plurality of voice signals within a voice communication resource (See FIG. 6) having a given bandwidth within a voice communication system. The method comprises the steps of subchanneling the voice communication resource (441, 442, 443) while placing a pair of the plurality of voice signals on a subchannel using single sideband modulation; modulating the pair of the plurality of voice signals about a pilot signal within the subchannel; and compressing the time of each of the voice signals within the plurality of subchannels, wherein these steps provide a compressed voice signal.--

END OF TRANSMISSION

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

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

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.

EXAMINER

ART UNIT	PAPER NUMBER
	13 B

DATE MAILED:

Please find below and/or attached an Office communication concerning this application or proceeding.

Commissioner of Patents and Trademarks

See Attachments

ELLIS B. RAMIREZ
PRIMARY EXAMINER
GROUP 2400

B

PTO-90C (Rev. 2/95)

☆U.S.GOVERNMENT PRINTING OFFICE 1996-411-516/40275

1- File Copy

Notice of Allowability	Application No. 08/764,656	Applicant(s) Leitch et al
	Examiner Demetra R. Smith	Group Art Unit 2414

All claims being allowable, PROSECUTION ON THE MERITS IS (OR REMAINS) CLOSED in this application. If not included herewith (or previously mailed), a Notice of Allowance and Issue Fee Due or other appropriate communication will be mailed in due course.

☒ This communication is responsive to *the Amendment of the claims*.

☒ The allowed claim(s) is/are *1, 4-14, 16, and 18-24*.

☐ The drawings filed on ______ are acceptable.

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

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

☐ received.

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

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

*Certified copies not received: ______.

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

A SHORTENED STATUTORY PERIOD FOR RESPONSE to comply with the requirements noted below is set to EXPIRE **THREE MONTHS** FROM THE "DATE MAILED" of this Office action. Failure to timely comply will result in ABANDONMENT of this application. Extensions of time may be obtained under the provisions of 37 CFR 1.136(a).

☐ Note the attached EXAMINER'S AMENDMENT or NOTICE OF INFORMAL APPLICATION, PTO-152, which discloses that the oath or declaration is deficient. A SUBSTITUTE OATH OR DECLARATION IS REQUIRED.

☐ Applicant MUST submit NEW FORMAL DRAWINGS

☐ because the originally filed drawings were declared by applicant to be informal.

☐ including changes required by the Notice of Draftsperson's Patent Drawing Review, PTO-948, attached hereto or to Paper No. ______.

☐ including changes required by the proposed drawing correction filed on ______, which has been approved by the examiner.

☐ including changes required by the attached Examiner's Amendment/Comment.

Identifying indicia such as the application number (see 37 CFR 1.84(c)) should be written on the reverse side of the drawings. The drawings should be filed as a separate paper with a transmittal lettter addressed to the Official Draftsperson.

☐ Note the attached Examiner's comment regarding REQUIREMENT FOR THE DEPOSIT OF BIOLOGICAL MATERIAL.

Any response to this letter should include, in the upper right hand corner, the APPLICATION NUMBER (SERIES CODE/SERIAL NUMBER). If applicant has received a Notice of Allowance and Issue Fee Due, the ISSUE BATCH NUMBER and DATE of the NOTICE OF ALLOWANCE should also be included.

Attachment(s)

☐ Notice of References Cited, PTO-892

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

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

☐ Notice of Informal Patent Application, PTO-152

☒ Interview Summary, PTO-413

☒ Examiner's Amendment/Comment

☐ Examiner's Comment Regarding Requirement for Deposit of Biological Material

☐ Examiner's Statement of Reasons for Allowance

EXAMINER'S AMENDMENT

1. An examiner's amendment to the record appears below. Should the changes and/or additions be unacceptable to applicant, an amendment may be filed as provided by 37 CFR 1.312. To ensure consideration of such an amendment, it MUST be submitted no later than the payment of the issue fee.

Authorization for this examiner's amendment was given in a telephone interview with James A. Lamb on June 12, 1997.

2. The application has been amended as follows:

In claim 1:

Replace "(a) subchanneling the voice communication resource and simultaneously placing at least one of each of the plurality of voice signals on a subchannel; and" with --(a) subchanneling the voice communication resource into a plurality of subchannels and simultaneously placing a pair of the plurality of voice signals on a subchannel;--.

Delete "(b) compressing the time of each...........provides a compressed voice signal."

Add --(b) modulating the pair of the plurality of voice signals about a pilot signal within the subchannel within the voice communication resource using single sideband modulation; and--.

Add --(c) compressing the time of each of the voice signals within the plurality of subchannels, wherein the result of steps (a), (b), and (c) provides a compressed voice signal.--

Cancel claims 2-3.

In claim 9:

Replace "(a) compressing the bandwidth of plurality of voice signals by subchanneling the voice communication resource and modulating each of the plurality of voice signals about a plurality of pilot signals within the voice communication resource;" with --(a) subchanneling the voice communication resource into a plurality of subchannels and simultaneously placing a pair of the plurality of voice signals on a subchannel;--.

Delete "(b)compressing the time of the voice signal,.......for transmission via a transmitter."

Add --(b) modulating the pair of the plurality of voice signals about a pilot signal within the subchannel within the voice communication resource using single sideband modulation; and--

Add --(c) compressing the time of each of the voice signals within the plurality of subchannels, wherein the result of steps (a), (b), and (c) provides a compressed voice signal for transmission via a transmitter.

In claim 12:

On line 9, insert --a pilot carrier signal generator that generates a pilot carrier for a pair of single side band signals which includes the processed signal, wherein the pilot carrier serves as an amplitude and phase reference for distortion that occurs as a result of channel aberrations; and-- after "a processing device for compressing the audio signal.....technique to provide a processed signal;" and before "a transmitter for transmitting the processed signal;".

On line 13, insert --a receiver circuit for detecting, filtering and responding to the amplitude and phase reference generated by the pilot carrier signal generator;-- after "a selective call receiver....processed signal" and before "a processing device for demodulating....reconstructed signal; and

Cancel claim 15

In claim 16:

Replace "a processing device for demodulating the received processed......to provide a reconstructed signal; and" with --a processing device for demodulating the received processed signal, wherein said processing device demodulates both and upper and a lower sideband of a subchannel, the upper and lower sidebands having independent information therein, and wherein said processing device uses time-scale expansion to provide a reconstructed signal;--

On line 9, insert --a receiver circuit for detecting, filtering and responding to the amplitude and phase reference generated by a pilot carrier signal generator in a transmitter at a base station; and-- after "a processing device for demodulating.....a reconstructed signal" and before "an amplifier for amplifying.....audio signal."

Cancel claim 17

In claim 23:

Replace "a digital signal processor for performing single sideband......the digitized received signal to provide a processed signal; and" with --a digital signal processor for performing single sideband demodulation of a subchannel having a pilot carrier and independent information on an upper and a lower sideband of a subchannel, wherein the digital signal processor also performs the functions of filtering the pilot carrier, performing automatic gain control using a feedforward loop, and decompanding the digitized received signal to provide a processed signal; and--

Cancel the Abstract as filed.

Replace the Abstract as follow:

-- The present invention comprises a method for compressing a plurality of voice signals within a voice communication resource (see FIG. 6) having a given bandwidth within a voice communication system (100). The method comprises the steps of subchanneling the voice communication resource into a plurality of subchannels (441, 442, 443), placing a pair of the plurality of voice signals (401, 402) on a subchannel (441); modulating the pair of the plurality of voice signals (401, 402) about a pilot signal (581) within the subchannel (441) using single sideband modulation; and compressing the time of each of the voice signals (401, 402) within the plurality of subchannels (441, 442, 443), wherein these step provide a compressed voice signal.--

3. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Demetra R. Smith whose telephone number is (703) 308-6989. The examiner can normally be reached on Tuesday-Friday from 8:00 to 5:00.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, E. Todd Voeltz, can be reached on (703) 305-9714. The fax phone number for this Group is (703) 305-9731.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the Group receptionist whose telephone number is (703) 305-3900.

Demetra R. Smith

Demetra R. Smith

June 12, 1997

UNITED STATES DEPARTMENT OF COMMERCE
Patent and Trademark Office

Address: Box ISSUE FEE
ASSISTANT COMMISSIONER FOR PATENTS
WASHINGTON, D.C. 20231

NOTICE OF ALLOWANCE AND ISSUE FEE DUE

APPLICATION NO.	FILING DATE	TOTAL CLAIMS	EXAMINER AND GROUP ART UNIT	DATE MAILED
First Named Applicant				

TITLE OF INVENTION

ATTY'S DOCKET NO.	CLASS-SUBCLASS	BATCH NO.	APPLN. TYPE	SMALL ENTITY	FEE DUE	DATE DUE

THE APPLICATION IDENTIFIED ABOVE HAS BEEN EXAMINED AND IS ALLOWED FOR ISSUANCE AS A PATENT. PROSECUTION ON THE MERITS IS CLOSED.

THE ISSUE FEE MUST BE PAID WITHIN THREE MONTHS FROM THE MAILING DATE OF THIS NOTICE OR THIS APPLICATION SHALL BE REGARDED AS ABANDONED. THIS STATUTORY PERIOD CANNOT BE EXTENDED.

HOW TO RESPOND TO THIS NOTICE:

I. Review the SMALL ENTITY status shown above. If the SMALL ENTITY is shown as yes, verify your current SMALL ENTITY status:

A. If the status is changed, pay twice the amount of the FEE DUE shown and notify the Patent and Trademark Office of the change in status, or

B. If the status is the same, pay the FEE DUE shown above.

If the SMALL ENTITY is shown as NO:

A. Pay FEE DUE shown above, or

B. File verified statementof Small Entity Status before, or with, payment of 1/2 the FEE DUE shown above.

II. Part B of this notice should be completed and returned to the Patent and Trademark Office (PTO) with your ISSUE FEE. Even if the ISSUE FEE has already been paid by charge to deposit account, Part B should be completed and returned. If you are charging the ISSUE FEE to your deposit account, section "6b" of Part B should be completed.

III. All communications regarding this application must give application number and batch number. Please direct all communication prior to issuance to Box ISSUE FEE unless advised to the contrary.

IMPORTANT REMINDER: 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.

PTOL-85 (REV. 05-96)(0651-0033)

3. PATENT AND TRADEMARK OFFICE COPY

File Copy

Notice of Allowability	Application No. 08/764,656	Applicant(s) Leitch et al
	Examiner Demetra R. Smith	Group Art Unit 2414

All claims being allowable, PROSECUTION ON THE MERITS IS (OR REMAINS) CLOSED in this application. If not included herewith (or previously mailed), a Notice of Allowance and Issue Fee Due or other appropriate communication will be mailed in due course.

☒ This communication is responsive to *the Amendment of the claims*.

☒ The allowed claim(s) is/are *1, 4-14, 16, and 18-24*.

☐ The drawings filed on ______ are acceptable.

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

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

☐ received.

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

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

*Certified copies not received: ______.

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

A SHORTENED STATUTORY PERIOD FOR RESPONSE to comply with the requirements noted below is set to EXPIRE **THREE MONTHS** FROM THE "DATE MAILED" of this Office action. Failure to timely comply will result in ABANDONMENT of this application. Extensions of time may be obtained under the provisions of 37 CFR 1.136(a).

☐ Note the attached EXAMINER'S AMENDMENT or NOTICE OF INFORMAL APPLICATION, PTO-152, which discloses that the oath or declaration is deficient. A SUBSTITUTE OATH OR DECLARATION IS REQUIRED.

☐ Applicant MUST submit NEW FORMAL DRAWINGS

☐ because the originally filed drawings were declared by applicant to be informal.

☐ including changes required by the Notice of Draftsperson's Patent Drawing Review, PTO-948, attached hereto or to Paper No. ______.

☐ including changes required by the proposed drawing correction filed on ______, which has been approved by the examiner.

☐ including changes required by the attached Examiner's Amendment/Comment.

Identifying indicia such as the application number (see 37 CFR 1.84(c)) should be written on the reverse side of the drawings. The drawings should be filed as a separate paper with a transmittal lettter addressed to the Official Draftsperson.

☐ Note the attached Examiner's comment regarding REQUIREMENT FOR THE DEPOSIT OF BIOLOGICAL MATERIAL.

Any response to this letter should include, in the upper right hand corner, the APPLICATION NUMBER (SERIES CODE/SERIAL NUMBER). If applicant has received a Notice of Allowance and Issue Fee Due, the ISSUE BATCH NUMBER and DATE of the NOTICE OF ALLOWANCE should also be included.

Attachment(s)

- ☐ Notice of References Cited, PTO-892
- ☐ Information Disclosure Statement(s), PTO-1449, Paper No(s). ______
- ☐ Notice of Draftsperson's Patent Drawing Review, PTO-948
- ☐ Notice of Informal Patent Application, PTO-152
- ☒ Interview Summary, PTO-413
- ☒ Examiner's Amendment/Comment
- ☐ Examiner's Comment Regarding Requirement for Deposit of Biological Material
- ☐ Examiner's Statement of Reasons for Allowance

EXAMINER'S AMENDMENT

1. An examiner's amendment to the record appears below. Should the changes and/or additions be unacceptable to applicant, an amendment may be filed as provided by 37 CFR 1.312. To ensure consideration of such an amendment, it MUST be submitted no later than the payment of the issue fee.

Authorization for this examiner's amendment was given in a telephone interview with James A. Lamb on June 12, 1997.

2. The application has been amended as follows:

In claim 1:

Replace "(a) subchanneling the voice communication resource and simultaneously placing at least one of each of the plurality of voice signals on a subchannel; and" with --(a) subchanneling the voice communication resource into a plurality of subchannels and simultaneously placing a pair of the plurality of voice signals on a subchannel;--.

Delete "(b) compressing the time of each...........provides a compressed voice signal."

Add --(b) modulating the pair of the plurality of voice signals about a pilot signal within the subchannel within the voice communication resource using single sideband modulation; and--.

32

[Add]--(c) compressing the time of each of the voice signals within the plurality of subchannels, wherein the result of steps (a), (b), and (c) provides a compressed voice signal.--

B2 concl.

Cancel claims 2-3.

In claim 9:

Replace "(a) compressing the bandwidth of plurality of voice signals by subchanneling the voice communication resource and modulating each of the plurality of voice signals about a plurality of pilot signals within the voice communication resource;" with--(a) subchanneling the voice communication resource into a plurality of subchannels and simultaneously placing a pair of the plurality of voice signals on a subchannel;--.

B3

Delete "(b)compressing the time of the voice signal,.......for transmission via a transmitter."

Add--(b) modulating the pair of the plurality of voice signals about a pilot signal within the subchannel within the voice communication resource using single sideband modulation; and--

B4

[Add]--(c) compressing the time of each of the voice signals within the plurality of subchannels, wherein the result of steps (a), (b), and (c) provides a compressed voice signal for transmission via a transmitter.

In claim 12:

50

B5 On line 9, insert --a pilot carrier signal generator that generates a pilot carrier for a pair of single side band signals which includes the processed signal, wherein the pilot carrier serves as an amplitude and phase reference for distortion that occurs as a result of channel aberrations; and-- after "a processing device for compressing the audio signal.....technique to provide a processed signal;" and before "a transmitter for transmitting the processed signal;".

B6 On line 13, insert --a receiver circuit for detecting, filtering and responding to the amplitude and phase reference generated by the pilot carrier signal generator;-- after "a selective call receiver....processed signal" and before "a processing device for demodulating....reconstructed signal; and

Cancel claim 15

In claim 16:

B7 Replace "a processing device for demodulating the received processed......to provide a reconstructed signal; and" with --a processing device for demodulating the received processed signal, wherein said processing device demodulates both and upper and a lower sideband of a subchannel, the upper and lower sidebands having independent information therein, and wherein said processing device uses time-scale expansion to provide a reconstructed signal;--

38 On line 9, insert --a receiver circuit for detecting, filtering and responding to the amplitude and phase reference generated by a pilot carrier signal generator in a

B3 Bed.

transmitter at a base station; and--after "a processing device for demodulating.....a reconstructed signal" and before "an amplifier for amplifying.....audio signal."

Cancel claim 17

In claim 23:

Replace "a digital signal processor for performing single sideband......the digitized received signal to provide a processed signal; and" with --a digital signal processor for performing single sideband demodulation of a subchannel having a pilot carrier and independent information on an upper and a lower sideband of a subchannel, wherein the digital signal processor also performs the functions of filtering the pilot carrier, performing automatic gain control using a feedforward loop, and decompanding the digitized received signal to provide a processed signal; and--

B9

Cancel the Abstract as filed.

Replace the Abstract as follow:

Abstract --The present invention comprises a method for compressing a plurality of voice signals within a voice communication resource (see FIG. 6) having a given bandwidth within a voice communication system (100). The method comprises the steps of subchanneling the voice communication resource into a plurality of subchannels (441, 442, 443), placing a pair of the plurality of voice signals (401, 402) on a subchannel (441); modulating the pair of the plurality of voice signals (401, 402) about a pilot signal (581) within the subchannel (441) using single

B10

sideband modulation; and compressing the time of each of the voice signals (401, 402) within the plurality of subchannels (441, 442, 443), wherein these step provide a compressed voice signal.--

3. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Demetra R. Smith whose telephone number is (703) 308-6989. The examiner can normally be reached on Tuesday-Friday from 8:00 to 5:00.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, E. Todd Voeltz, can be reached on (703) 305-9714. The fax phone number for this Group is (703) 305-9731.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the Group receptionist whose telephone number is (703) 305-3900.

Demetra R. Smith
Demetra R. Smith

June 12, 1997

UNITED STATES DEPARTMENT OF COMMERCE
Patent and Trademark Office

Address: Box ISSUE FEE
ASSISTANT COMMISSIONER FOR PATENTS
WASHINGTON, D.C. 20231

NOTICE OF ALLOWANCE AND ISSUE FEE DUE

APPLICATION NO.	FILING DATE	TOTAL CLAIMS	EXAMINER AND GROUP ART UNIT	DATE MAILED

First Named Applicant

TITLE OF INVENTION

ATTY'S DOCKET NO.	CLASS-SUBCLASS	BATCH NO.	APPLN. TYPE	SMALL ENTITY	FEE DUE	DATE DUE

THE APPLICATION IDENTIFIED ABOVE HAS BEEN EXAMINED AND IS ALLOWED FOR ISSUANCE AS A PATENT. PROSECUTION ON THE MERITS IS CLOSED.

THE ISSUE FEE MUST BE PAID WITHIN THREE MONTHS FROM THE MAILING DATE OF THIS NOTICE OR THIS APPLICATION SHALL BE REGARDED AS ABANDONED. THIS STATUTORY PERIOD CANNOT BE EXTENDED.

HOW TO RESPOND TO THIS NOTICE:

I. Review the SMALL ENTITY status shown above. If the SMALL ENTITY is shown as yes, verify your current SMALL ENTITY status:

A. If the status is changed, pay twice the amount of the FEE DUE shown and notify the Patent and Trademark Office of the change in status, or

B. If the status is the same, pay the FEE DUE shown above.

If the SMALL ENTITY is shown as NO:

A. Pay FEE DUE shown above, or

B. File verified statementof Small Entity Status before, or with, payment of 1/2 the FEE DUE shown above.

II. Part B of this notice should be completed and returned to the Patent and Trademark Office (PTO) with your ISSUE FEE. Even if the ISSUE FEE has already been paid by charge to deposit account, Part B should be completed and returned. If you are charging the ISSUE FEE to your deposit account, section "6b" of Part B should be completed.

III. All communications regarding this application must give application number and batch number. Please direct all communication prior to issuance to Box ISSUE FEE unless advised to the contrary.

IMPORTANT REMINDER: 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.

PTOL-85 (REV. 05-98)(0651-0033)

2. YOUR COPY

FAX RECEIVED

JUN 1 2 1997

GROUP 2300

OFFICIAL

INTELLECTUAL PROPERTY DEPARTMENT

MOTOROLA INC.

PAGING AND TELEPOINT SYSTEMS GROUP
BOYNTON BEACH, FLORIDA 33426

FAX

MESSAGE TRANSMITTAL

FAX: 561-739-2825
VERIFY 561-739-2860

To: Examiner Demetra Smith

Phone: (703) 308-6989
FAX: (703)-305-9731
Application number: 08/764,656

From: James A. Lamb

Agent: 38, 529
Phone: 407-739-2862

Number of Pages: 8

June 12, 1997

FAX RECEIVED

JUN 1 2 1997

GROUP 2300

Examiner Smith,

In accordance with our telephone conversation this afternoon, I am transmitting herewith a set of claims corresponding to the amended set of claims transmitted to you on May 22, but showing the additions and deletions with respect the US amended claims, using brackets and underlines. I have also changed the claim numbers to the U.S. claim numbers. The added phrase "using single side band modulation" has been moved from step (a) to step (b) in claims 1 and 9 to improve clarity. Changes have been made to the Abstract for improved clarity. No new matter has been added.

Sincerely,

James A. Lamb

James A. Lamb

NOTICE: This facsimile transmission may contain information that is confidential, privileged or exempt from disclosure under applicable law. It is intended only for the person to whom it is addressed. Unauthorized use, disclosure, copying or distribution may expose you to legal liability. If you have received this transmission in error, please immediately notify us by telephone (collect) to arrange for return of the documents received and any copies made. Thank you.

In the claims.

Please replace the claims as filed with the following claims:

1. (Amended) A method for compressing a plurality of voice signals within a voice communication resource having a given bandwidth within a voice communication system, comprising the steps of:

(a) subchanneling the voice communication resource into a plurality of subchannels and simultaneously placing [at least one of each of] a pair of the plurality of voice signals on a subchannel; [and]

(b) modulating the pair of the plurality of voice signals about a pilot signal within the subchannel within the voice communication resource using single sideband modulation; and

[(b)](c) compressing the time of each of the voice signals within [each of] the plurality of subchannels, wherein the result of steps (a), [and] (b), and (c) provides a compressed voice signal.

Claims 2-3 are cancelled.

4. The method of claim 1, wherein the step of subchanneling further comprises the step of using quadrature amplitude modulation.

5. The method of claim 1, wherein the step of compressing the time of each of the voice signals further comprises the step using time-scale compression on the voice signals.

6. The method of claim 1, wherein the step of compressing the time of each of the voice signals further comprises the step of using Waveform Similarity based Overlap-Add (WSOLA) time compression on the voice signals.

7. The method of claim 1, wherein the step of compressing the time of each of the voice signals further comprises the speaker dependent steps of identifying pitch periods within each of the voice signals and transmitting data from one pitch period to alter a time-scaling factor.

8. The method of claim 1, wherein the step of compressing the time of each of the voice signals comprises the step of using a speaker dependent modification of the Waveform Similarity based Overlap-Add (WSOLA) time compression technique on the voice signals.

9. (Amended) A method for compressing a plurality of voice signals within a voice communication resource within a voice communication system, comprising the steps of:

(a) [compressing the bandwidth of plurality of voice signals by] subchanneling the voice communication resource [and modulating each of the plurality of voice signals about a plurality of pilot signals within the voice communication resource] into a plurality of subchannels and simultaneously placing a pair of the plurality of voice signals on a subchannel;

(b) modulating the pair of the plurality of voice signals about a pilot signal within the subchannel within the voice communication resource using single sideband modulation; and

[(b)] (c) compressing the time of each of the voice signals within the plurality of subchannels, wherein the result of steps (a), [and] (b), and (c) provides a compressed voice signal for transmission via a transmitter.

10. The method of claim 9, wherein the method further comprises the step at the transmitter of transmitting the compressed voice signal to a plurality of selective call receivers.

11. The method of claim 9, wherein the method further comprises the step of receiving the compressed voice signal and demodulating the compressed bandwidth signals at one of the the plurality of selective call receivers.

SENT BY:MOTOROLA PAGING ; 6-12-97 ; 8:45AM ; BOYNTON→ 17033058731;# 3/ 8

12. (Amended) A communication system using voice compression having at least one transmitter base station and a plurality of selective call receivers, comprising:

at the transmitter base station:

an input device for receiving an audio signal;

a processing device for compressing the audio signal using time-scale compression and a single side band modulation technique to provide a processed signal;

a pilot carrier signal generator that generates a pilot carrier for a pair of single side band signals which includes the processed signal, wherein the pilot carrier serves as an amplitude and phase reference for distortion that occurs as a result of channel aberrations; and

a transmitter for transmitting the processed signal;

at each of the plurality of selective call receivers:

a selective call receiver for receiving the transmitted processed signal;

a receiver circuit for detecting, filtering and responding to the amplitude and phase reference generated by the pilot carrier signal generator;

a processing device for demodulating the received processed signal using single side band demodulation and time-scale expansion to provide a reconstructed signal; and

an amplifier for amplifying the reconstructed signal into an reconstructed audio signal.

13. The communication system of claim 10, wherein the single sideband modulation technique provides for the transmission of a single message split between an upper sideband and a lower sideband.

14. The communication system of claim 10, wherein the single sideband modulation technique provides for the transmission of a single message repeated on an upper sideband and lower sideband.

Claim 15 is canceled.

16. (Amended) A selective call receiver for receiving compressed voice signals, comprising:

a selective call receiver for receiving a transmitted processed signal that includes compressed voice signals that have been compressed using time-scale compression;

a processing device for demodulating the received processed signal [using single sideband demodulation and], wherein said processing device demodulates both an upper and a lower sideband of a subchannel, the upper and lower sidebands having independent information therein, and wherein said processing device uses time-scale expansion to provide a reconstructed signal;

a receiver circuit for detecting, filtering and responding to the amplitude and phase reference generated by a pilot carrier signal generator in a transmitter at a base station; and

an amplifier for amplifying the reconstructed signal into an reconstructed audio signal.

Claim 17 is cancelled.

18. A selective call paging base station for transmitting selective call signals on a communication resource having a predetermined bandwidth, comprising:

an input device for receiving a plurality of audio signals;

a means for subchannelizing the communication resource into a predetermined number of subchannels;

an amplitude compression and filtering module for each subchannel for compressing the amplitude of the respective audio signal and filtering the respective audio signal ;

a time compression module for compressing the time of the respective audio signal for each subchannel; and

a quadrature amplitude modulation transmitter for transmitting the processed signal.

19. The selective call paging base station of claim 18, wherein the input device for receiving a plurality of audio signals comprises a paging terminal for receiving phone messages or data messages from a computing device.

20. The selective call paging base station of claim 18, wherein the amplitude compression and filtering module comprises an anti-alias filter coupled to an analog-to-digital converter coupled to a band-pass filter coupled to an automatic gain controller and clipper circuit.

21. The selective call paging base station of claim 18, wherein the time compression module comprises a processing device for compressing the audio signal using a time-scale compression technique.

22. The selective call paging base station of claim 18, wherein the time compression module comprises a processing device for compressing the audio signal using a WSOLA time compression technique.

23. (Amended) A selective call receiver unit for receiving compressed voice selective call signals, comprising:

a receiver having a analog to digital converter for providing a digitized received signal;

a digital signal processor for performing single sideband demodulation [and at least one of] of a subchannel having a pilot carrier and independent information on an upper and a lower sideband of a subchannel, wherein the digital signal processor also performs the functions of filtering [a] the pilot carrier, performing automatic gain control using a feedforward loop, and decompanding the digitized received signal to provide a processed signal; and

a digital to analog converter and reconstruction filter for converting the processed signal into a digitized audio signal; and

an amplifier for amplifying the digitized audio signal.

24. A communication base station, comprising:

a terminal for receiving an audio speech signal;

an analog to digital converter for converting the audio speech signal into a digitized speech signal;

a digital signal processor for processing the digitized speech signal by performing the function of splitting the digitized speech signal and at least one of the functions of bandpass filtering, automatic gain control , time scaling, companding, or buffering; and

a transmitter having at least a Hilbert transform filter coupled to a digital to analog converter coupled to a reconstruction filter coupled to a quadrature amplitude modulator which is coupled to a radio frequency power amplifier. --

In the Abstract

Please cancel the Abstract as filed and use the following abstract:

--The present invention comprises a method for compressing a plurality of voice signals within a voice communication resource (see FIG. 6) having a given bandwidth within a voice communication system (100). The method comprises the steps of subchanneling the voice communication resource into a plurality of subchannels (441, 442, 443), placing a pair of the plurality of voice signals (401, 402) on a subchannel (441); modulating the pair of the plurality of voice signals (401, 402) about a pilot signal (581) within the subchannel (441) using single sideband modulation; and compressing the time of each of the voice signals (401, 402) within the plurality of subchannels (441, 442, 443), wherein these steps provide a compressed voice signal.--

END OF TRANSMISSION

Interview Summary	Application No. 08/764,656	Applicant(s) Leitch et al
	Examiner Demetra R. Smith	Group Art Unit 2414

All participants (applicant, applicant's representative, PTO personnel):

(1) *Demetra R. Smith* (3) ______

(2) *James A. Lamb* (4) ______

Date of Interview *Jun 12, 1997*

Type: ☒ Telephonic ☐ Personal (copy is given to ☐ applicant ☐ applicant's representative).

Exhibit shown or demonstration conducted: ☐ Yes ☒ No. If yes, brief description:

Agreement ☒ was reached. ☐ was not reached.

Claim(s) discussed: *1-3, 9, 12, 15-17, and 23*

Identification of prior art discussed:
None

Description of the general nature of what was agreed to if an agreement was reached, or any other comments:
The applicant filed a PCT, in which the claims were amended. The applicant wanted the claims and the abstract of the parent case to correspond with the filed PCT. The attorney/agent agreed upon an examiner's amendment to the claims and the abstract of the parent case.

(A fuller description, if necessary, and a copy of the amendments, if available, which the examiner agreed would render the claims allowable must be attached. Also, where no copy of the amendents which would render the claims allowable is available, a summary thereof must be attached.)

1. ☒ It is not necessary for applicant to provide a separate record of the substance of the interview.

Unless the paragraph above has been checked to indicate to the contrary, A FORMAL WRITTEN RESPONSE TO THE LAST OFFICE ACTION IS NOT WAIVED AND MUST INCLUDE THE SUBSTANCE OF THE INTERVIEW. (See MPEP Section 713.04). If a response to the last Office action has already been filed, APPLICANT IS GIVEN ONE MONTH FROM THIS INTERVIEW DATE TO FILE A STATEMENT OF THE SUBSTANCE OF THE INTERVIEW.

2. ☐ Since the Examiner's interview summary above (including any attachments) reflects a complete response to each of the objections, rejections and requirements that may be present in the last Office action, and since the claims are now allowable, this completed form is considered to fulfill the response requirements of the last Office action. Applicant is not relieved from providing a separate record of the interview unless box 1 above is also checked.

Examiner Note: You must sign and stamp this form unless it is an attachment to a signed Office action.

Interview Summary	Application No. 08/764,656	Applicant(s) Leitch et al
	Examiner Demetra R. Smith	Group Art Unit 2414

All participants (applicant, applicant's representative, PTO personnel):

(1) *Demetra R. Smith* (3) ______

(2) *James A. Lamb* (4) ______

Date of Interview *Jun 12, 1997*

Type: ☒ Telephonic ☐ Personal (copy is given to ☐ applicant ☐ applicant's representative).

Exhibit shown or demonstration conducted: ☐ Yes ☒ No. If yes, brief description:

Agreement ☒ was reached. ☐ was not reached.

Claim(s) discussed: *1-3, 9, 12, 15-17, and 23*

Identification of prior art discussed:
None

Description of the general nature of what was agreed to if an agreement was reached, or any other comments:
The applicant filed a PCT, in which the claims were amended. The applicant wanted the claims and the abstract of the parent case to correspond with the filed PCT. The attorney/agent agreed upon an examiner's amendment to the claims and the abstract of the parent case.

(A fuller description, if necessary, and a copy of the amendments, if available, which the examiner agreed would render the claims allowable must be attached. Also, where no copy of the amendents which would render the claims allowable is available, a summary thereof must be attached.)

1. ☒ It is not necessary for applicant to provide a separate record of the substance of the interview.

Unless the paragraph above has been checked to indicate to the contrary, A FORMAL WRITTEN RESPONSE TO THE LAST OFFICE ACTION IS NOT WAIVED AND MUST INCLUDE THE SUBSTANCE OF THE INTERVIEW. (See MPEP Section 713.04). If a response to the last Office action has already been filed, APPLICANT IS GIVEN ONE MONTH FROM THIS INTERVIEW DATE TO FILE A STATEMENT OF THE SUBSTANCE OF THE INTERVIEW.

2. ☐ Since the Examiner's interview summary above (including any attachments) reflects a complete response to each of the objections, rejections and requirements that may be present in the last Office action, and since the claims are now allowable, this completed form is considered to fulfill the response requirements of the last Office action. Applicant is not relieved from providing a separate record of the interview unless box 1 above is also checked.

Examiner Note: You must sign and stamp this form unless it is an attachment to a signed Office action.

MAIL ROOM JUN 1997

PART B—ISSUE FEE TRANSMITTAL

EXPRESS MAIL #EM450789502US

B

MAILING INSTRUCTIONS: This form should be used for transmitting the ISSUE FEE. Blocks 2 through 6 should be completed where appropriate. All further correspondence including the Issue Fee Receipt, the Patent, advance orders and notification of maintenance fees will be mailed to addresses entered in Block 1 unless you direct otherwise, by: (a) specifying a new correspondence address in Block 3 below; or (b) providing the PTO with a separate "FEE ADDRESS" for maintenance fee notifications with the payment of Issue Fee or thereafter. See reverse for Certificate of Mailing, below.

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.

Burden Hour Statement: This form is estimated to take 0.2 hours to complete. Time will vary depending on the needs of the individual case. Any comments on the amount of time required to complete this form should be sent to the Chief Information Officer, Patent and Trademark Office, Washington, D.C. 20231.
DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. SEND TO: Box Issue Fee, Assistant Commissioner for Patents, Washington D.C. 20231

1. CORRESPONDENCE ADDRESS

B3M1/0319

MOTOROLA INC
IP LAW DEPARTMENT MS 96
1500 GATEWAY BLVD
BOYNTON BEACH FL 33426-8753

MAIL ROOM JUN 16 1997 PAT. & TRADEMARK OFF.

2. INVENTOR(S) ADDRESS CHANGE (Complete only if there is a change)

INVENTOR'S NAME

Street Address

City, State and Zip Code

CO-INVENTOR'S NAME

Street Address

City, State and Zip Code

☐ Check if additional changes are enclosed

APPLICATION NO.	FILING DATE	TOTAL CLAIMS	EXAMINER AND GROUP ART UNIT		DATE MAILED
08/764,656	12/11/96	024	SMITH, D	2414	03/19/97

First Named Applicant: LEITCH, CLIFFORD D.

TITLE OF INVENTION: VOICE COMPRESSION METHOD AND APPARATUS IN A COMMUNICATION SYSTEM

ATTY'S DOCKET NO.		CLASS-SUBCLASS	BATCH NO.	APPLN. TYPE	SMALL ENTITY	FEE DUE	DATE DUE
3	PT00600UC01	364-514.00R	K90	UTILITY	NO	$1290.00	06/19/97

3. Correspondence address change (Complete only if there is a change)

08/19/1997 LBERGER 00000057 DAH:134778 08764656
01 FC:142 1290.00 CH
02 FC:561 60.00 CH

4. For printing on the patent front page, list the names of not more than 3 registered patent attorneys or agents OR, alternatively, the name of a firm having as a member a registered attorney or agent. If no name is listed, no name will be printed.

1 James A. Lamb

2

3

5. ASSIGNMENT DATA TO BE PRINTED ON THE PATENT (print or type)

(1) NAME OF ASSIGNEE: MOTOROLA, INC.

(2) ADDRESS: (CITY & STATE OR COUNTRY) SCHAUMBURG, ILLINOIS

A. ☐ This application is NOT assigned.

☒ Assignment previously submitted to the Patent and Trademark Office.

☐ Assignment is being submitted under separate cover. Assignment should be directed to Box ASSIGNMENTS.

PLEASE NOTE: Unless an assignee is identified in Block 5, no assignee data will appear on the patent. Inclusion of assignee data is only appropriate when an assignment has been perviously submitted to the PTO or is being submitted under separate cover. Completion of this form is NOT a substitute for filing an assignment.

6a. The following fees are enclosed:
☐ Issue Fee ☐ Advance Order - # of Copies ______

6b. The following fees should be charged to:
DEPOSIT ACCOUNT NUMBER 13-4778
(ENCLOSE A COPY OF THIS FORM)
☒ Issue Fee ☒ Advance Order - # of Copies 20
☐ Any Deficiencies in Enclosed Fees

The COMMISSIONER OF PATENTS AND TRADEMARKS is requested to apply the Issue Fee to the application identified above.

(Authorized Signature) James A. Lamb (Date) 6/16/97

NOTE: The Issue Fee will not be accepted from anyone other than the applicant; a registered attorney or agent; or the assignee or other party in interest as shown by the records of the Patent and Trademark Office.

Certificate of Mailing

Note: If this certificate of mailing is used, it can be used to transmit the Issue Fee. 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.

I hereby certify that this correspondence is being deposited with the United States Postal Service with sufficent postage as first class mail in an envelope addressed to: **Box ISSUE FEE**
Assistant Commissioner for Patents
Washington, D.C. 20231

on: June 16, 1997 (Date)
Cathy L. Tredinnick (Name of person making deposit)
Cathy Tredinnick (Signature)
6/16/97 (Date)

1. TRANSMIT THIS FORM WITH FEE

PTOL-85B (REV. 05-96) Approved for use through 06/31/99. OMB 0651-0033

Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE

PART B—ISSUE FEE TRANSMITTAL

MAILING INSTRUCTIONS: This form should be used for transmitting the ISSUE FEE. Blocks 2 through 6 should be completed where appropriate. All futher correspondence including the Issue Fee Receipt, the Patent, advance orders and notification of maintenance fees will be mailed to addresses entered in Block 1 unless you direct otherwise, by: (a) specifyinga new correspondence address in Block 3 below; or (b) providing the PTO with a separate "FEE ADDRESS" for maintenance fee notifications with the payment of Issue Fee or thereafter. **See reverse for Certificate of Mailing, below.**

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.

Burden Hour Statement: This form is estimated to take 0.2 hours to complete. Time will vary depending on the needs of the individual case. Any comments on the amount of time required to complete this form should be sent to the Chief Information Officer, Patent and Trademark Office, Washington, D.C. 20231.
DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. SEND TO: Box Issue Fee, Assistant Commissioner for Patents, Washington D.C. 20231

1. CORRESPONDENCE ADDRESS

2. INVENTOR(S) ADDRESS CHANGE (Complete only if there is a change)

INVENTOR'S NAME

Street Address

City, State and Zip Code

CO-INVENTOR'S NAME

Street Address

City, State and Zip Code

☐ Check if additional changes are enclosed

APPLICATION NO.	FILING DATE	TOTAL CLAIMS	EXAMINER AND GROUP ART UNIT	DATE MAILED

First Named Applicant

TITLE OF INVENTION

ATTY'S DOCKET NO.	CLASS-SUBCLASS	BATCH NO.	APPLN. TYPE	SMALL ENTITY	FEE DUE	DATE DUE

3. Correspondence address change (Complete only if there is a change)

4. For printing on the patent front page, list the names of not more than 3 registered patent attorneys or agents OR, alternatively, the name of a firm having as a member a registered attorney or agent. If no name is listed, no name will be printed.

1 ______

2 ______

3 ______

5. ASSIGNMENT DATA TO BE PRINTED ON THE PATENT (print or type)

(1) NAME OF ASSIGNEE:

(2) ADDRESS: (CITY & STATE OR COUNTRY)

A. ☐ This application is NOT assigned.

☐ Assignment previously submitted to the Patent and Trademark Office.

☐ Assignment is being submitted under separate cover. Assignment should be directed to Box ASSIGNMENTS.

PLEASE NOTE: Unless an assignee is identified in Block 5, no assignee data will appear on the patent. Inclusion of assignee data is only appropriate when an assignment has been perviously submitted to the PTO or is being submitted under separate cover. Completion of this form is NOT a substitute for filing an assignment.

6a. The following fees are enclosed:
☐ Issue Fee ☐ Advance Order - # of Copies ______

6b. The following fees should be charged to:
DEPOSIT ACCOUNT NUMBER ______
(ENCLOSE A COPY OF THIS FORM)
☐ Issue Fee ☐ Advance Order - # of Copies ______
☐ Any Deficiencies in Enclosed Fees

The COMMISSIONER OF PATENTS AND TRADEMARKS is requested to apply the Issue Fee to the application identified above.

(Authorized Signature) (Date)

NOTE: The Issue Fee will not be accepted from anyone other than the applicant; a registered attorney or agent; or the assignee or other party in interest as shown by the records of the Patent and Trademark Office.

Certificate of Mailing

Note: If this certificate of mailing is used, it can be used to transmit the Issue Fee. 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.

I hereby certify that this correspondence is being deposited with the United States Postal Service with sufficent postage as first class mail in an envelope addressed to: **Box ISSUE FEE**
Assistant Commissioner for Patents
Washington, D.C. 20231

on: ______ **(Date)**

______ **(Name of person making deposit)**

______ **(Signature)**

______ **(Date)**

PTOL-85B (REV. 05-98) Approved for use through 05/31/99. OMB 0651-0033 **1. TRANSMIT THIS FORM WITH FEE** Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE

[redacted]

Transaction History Date ____________

Date information retrieved from USPTO Patent Application Information Retrieval (PAIR) system records at www.uspto.gov

PTO UTILITY GRANT

Paper Number 14

The United States of America

The Commissioner of Patents and Trademarks

Has received an application for a patent for a new and useful invention. The title and description of the invention are enclosed. The requirements of law have been complied with, and it has been determined that a patent on the invention shall be granted under the law.

Therefore, this

United States Patent

Grants to the person(s) having title to this patent the right to exclude others from making, using, offering for sale, or selling the invention throughout the United States of America or importing the invention into the United States of America for the term set forth below, subject to the payment of maintenance fees as provided by law.

If this application was filed prior to June 8, 1995, the term of this patent is the longer of seventeen years from the date of grant of this patent or twenty years from the earliest effective U.S. filing date of the application, subject to any statutory extension.

If this application was filed on or after June 8, 1995, the term of this patent is twenty years from the U.S. filing date, subject to an statutory extension. If the application contains a specific reference to an earlier filed application or applications under 35 U.S.C. 120, 121 or 365(c), the term of the patent is twenty years from the date on which the earliest application was filed, subject to any statutory extension.

Bruce Lehman

Commissioner of Patents and Trademarks

Sandra J Morton

Attest

Form PTO-1584 (Rev. 2/97)

(RIGHT INSIDE)

FPI-LOM

UNITED STATES PAT[illegible]T AND TRADEMARK OFFICE

Commissioner for Patents
United States Patent and Trademark Office
P.O. Box 1450
Alexandria, VA 22313-1450
www.uspto.gov

P75M

MOTOROLA INC
IP LAW DEPARTMENT MS 96
1500 GATEWAY BLVD
BOYNTON BEACH FL 33426-8753

DATE PRINTED

05/25/09

MAINTENANCE FEE REMINDER

According to the records of the U.S. Patent and Trademark Office (USPTO) the maintenance fee for the patent(s) listed below (for which the above address is on record as the fee address under 37 CFR 1.363) has not been paid within the six-month period set forth in 37 CFR 1.362(d). THE MAINTENANCE FEE MAY STILL BE PAID WITH THE APPLICABLE SURCHARGE SET FORTH IN 37 CFR 1.20(h), WITHIN THE SIX-MONTH GRACE PERIOD SET FORTH IN 37 CFR 1.362(e).

Unless payment of the maintenance fee and the applicable surcharge is received in the USPTO within the six-month grace period, THE PATENT WILL EXPIRE AS OF THE END OF THE GRACE PERIOD. 35 U.S.C. 41(b).

The total payment due is the amount required on the date the fee is paid (and not necessarily the amount indicated below). All USPTO fees (including maintenance fees) are subject to change. Customers should refer to the USPTO Web site (www.uspto.gov) or call the Maintenance Fee Branch at 571-272-6500 for the most current fee amounts for the correct entity status before submitting payment. The total payment due indicated below is based on the entity status according to current Office records (shown below).

Timely payment of the total payment due is required in order to avoid expiration of the patent. A maintenance fee payment can be timely made using the certificate of mailing or transmission procedure set forth in 37 CFR 1.8.

PATENT NUMBER	FEE MAINT. AMT	SURCHG	U.S. APPL NUMBER	PATENT ISSUE DATE	APPL. FILING DATE	PAY-MENT YEAR	SMALL ENTITY?	TOTAL PYMT DUE	ATTORNEY DOCKET NUMBER
5689440	4110	130	08764656	11/18/97	12/11/96	12	NO	4240	PT00600UC01

The maintenance fee and the applicable surcharge can be paid quickly and easily over the Internet at www.uspto.gov by electronic funds transfer (EFT), credit card, or USPTO deposit account payment methods. The mailing address for all maintenance fee payments <u>not electronically submitted over the Internet</u> is: U.S. Patent and Trademark Office, P.O. Box 979070, St. Louis, MO 63197-9000.

Direct any questions about this notice to: Mail Stop M Correspondence, Director of the United States Patent and Trademark Office, P.O. Box 1450, Alexandria, VA 22313-1450.

NOTE: This notice was automatically generated based on the amount of time that elapsed since the date a patent was granted. It is possible that the patent term may have ended or been shortened due to a terminal disclaimer that was filed in the application. Also, for any patent that issued from an application filed on or after June 8, 1995 containing a specific reference to in earlier filed application or applications under 35 U.S.C. 120, 121, or 365(c), the patent erm ends 20 years from the date on which the earliest such application was filed, unless ie term was adjusted or extended under 35 U.S.C. 154 or 156. Patentee should determine the levant patent term for a patent before paying the maintenance fee.

401 (7/2007)

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

P75M

MOTOROLA INC
IP LAW DEPARTMENT MS 96
1500 GATEWAY BLVD
BOYNTON BEACH FL 33426-8753

DATE PRINTED

12/14/09

NOTICE OF PATENT EXPIRATION

According to the records of the U.S. Patent and Trademark Office (USPTO), payment of the maintenance fee for the patent(s) listed below has not been received timely prior to the end of the six-month grace period in accordance with 37 CFR 1.362(e). THE PATENT(S) LISTED BELOW HAS THEREFORE EXPIRED AS OF THE END OF THE GRACE PERIOD. 35 U.S.C. 41(b). Notice of the expiration will be published in the USPTO Official Gazette.

Expired patents may be reinstated in accordance with 37 CFR 1.378 if upon petition, the maintenance fee and the surcharge set forth in 37 CFR 1.20(i) are paid, AND the delay in payment of the maintenance fee is shown to the satisfaction of the Director to have been unavoidable or unintentional. 35 U.S.C. 41(c)(1).

If the Director accepts payment of the maintenance fee and surcharge upon petition under 37 CFR 1.378, the patent shall be considered as not having expired but would be subject to the intervening rights and conditions set forth in 35 U.S.C. 41(c)(2).

For instructions on filing a petition under 37 CFR 1.378 to reinstate an expired patent, customers should call the Office of Petitions Help Desk at 571-272-3282 or refer to the USPTO Web site at www.uspto.gov/web/offices/pac/dapp/petitionspractice.html. The USPTO also permits reinstatement under 37 CFR 1.378(c) by electronic petition (e-petition) using EFS-Web; e-petitions may be automatically granted if all the eligibility requirements are met. For further information on filing an e-petition, please call the Electronic Business Center (EBC) at 866-217-9197 (toll-free) or 571-272-4100 or refer to the EBC's e-petition guide at www.uspto.gov/ebc/portal/efs/petition_quickstart.pdf.

PATENT NUMBER	U.S. APPLICATION NUMBER	PATENT ISSUE DATE	APPLICATION FILING DATE	EXPIRATION DATE	ATTORNEY DOCKET NUMBER
5689440	08764656	11/18/97	12/11/96	11/18/09	PT00600UC01

NOTE: This notice was automatically generated based on the amount of time that elapsed since the date a patent was granted. It is possible that the patent term may have ended or been shortened due to a terminal disclaimer that was filed in the application. Also, for any patent that issued from an application filed on or after June 8, 1995 containing a specific reference to an earlier filed application or applications under 35 U.S.C. 120, 121, or 365(c), the patent term ends 20 years from the date on which the earliest such application was filed, unless the term was adjusted or extended under 35 U.S.C. 154 or 156.

MA441D (11/2008)

ID	MCH	TPE	NAME OR ACCOUNT	C-NBR	MLEDTE	CURDTE	F-C	$ AMOUNT
D	290	1	134778	00080	961210	970108	101	1,200.00

NO MORE TRANSACTIONS

END OF YOUR QUERY

CREDIT

PATENT APPLICATION FEE DETERMINATION RECORD
Effective October 1, 1996

Application or Docket Number: 764656

CLAIMS AS FILED - PART I

FOR	(Column 1) NUMBER FILED	(Column 2) NUMBER EXTRA	SMALL ENTITY RATE	SMALL ENTITY FEE	OR	OTHER THAN SMALL ENTITY RATE	OTHER THAN SMALL ENTITY FEE
BASIC FEE				385.00	OR		770.00
TOTAL CLAIMS	24 minus 20=	* 4	X$11=		OR	X$22=	88
INDEPENDENT CLAIMS	7 minus 3 =	* 4	X40=		OR	X80=	320
MULTIPLE DEPENDENT CLAIM PRESENT			+130=		OR	+260=	
			TOTAL		OR	TOTAL	1176

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

CLAIMS AS AMENDED - PART II

AMENDMENT A

	(Column 1) CLAIMS REMAINING AFTER AMENDMENT		(Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR	(Column 3) PRESENT EXTRA	SMALL ENTITY RATE	ADDITIONAL FEE	OR	OTHER THAN SMALL ENTITY RATE	ADDITIONAL FEE
Total	* 24	Minus	** 24	= —	X$11=		OR	X$22=	
Independent	* 7	Minus	*** 7	= —	X40=		OR	X80=	
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM					+130=		OR	+260=	
					TOTAL ADDIT. FEE		OR	TOTAL ADDIT. FEE	

AMENDMENT B

	(Column 1) CLAIMS REMAINING AFTER AMENDMENT		(Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR	(Column 3) PRESENT EXTRA	RATE	ADDITIONAL FEE	OR	RATE	ADDITIONAL FEE
Total	*	Minus	**	=	X$11=		OR	X$22=	
Independent	*	Minus	***	=	X40=		OR	X80=	
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM					+130=		OR	+260=	
					TOTAL ADDIT. FEE		OR	TOTAL ADDIT. FEE	

AMENDMENT C

	(Column 1) CLAIMS REMAINING AFTER AMENDMENT		(Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR	(Column 3) PRESENT EXTRA	RATE	ADDITIONAL FEE	OR	RATE	ADDITIONAL FEE
Total	*	Minus	**	=	X$11=		OR	X$22=	
Independent	*	Minus	***	=	X40=		OR	X80=	
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM					+130=		OR	+260=	
					TOTAL ADDIT. FEE		OR	TOTAL ADDIT. FEE	

* If the entry in column 1 is less than the entry in column 2, write "0" in column 3.
** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 20, enter "20."
*** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 3, enter "3."
The "Highest Number Previously Paid For" (Total or Independent) is the highest number found in the appropriate box in column 1.

FORM PTO-875
(Rev. 8/96)

Patent and Trademark Office, U.S. DEPARTMENT OF COMMERCE

Form PTO 1130
(REV 2/94)

U.S. DEPARTMENT OF COMMERCE
Patent and Trademark Office

PACE DATA ENTRY CODING SHEET

1ST EXAMINER	Jay	DATE	1/13/97
2ND EXAMINER		DATE	

APPLICATION NUMBER	TYPE APPL	FILING DATE (MONTH DAY YEAR)	SPECIAL HANDLING	GROUP ART UNIT	CLASS	SHEETS OF DRAWING
08/764656	2	12 11 94	0	2414	364	14

TOTAL CLAIMS	INDEPENDENT CLAIMS	SMALL ENTITY?	FILING FEE	FOREIGN LICENSE	ATTORNEY DOCKET NUMBER
24	7	0	1178	y	PT0060004-C01

CONTINUITY DATA

CONT CODE	STATUS CODE	PARENT APPLICATION SERIAL NUMBER	PCT APPLICATION SERIAL NUMBER	PARENT PATENT NUMBER	PARENT FILING DATE (MONTH DAY YEAR)
02	3	08395747	PCT/ /		02 28 95
			PCT/ /		
			PCT/ /		
			PCT/ /		
			PCT/ /		

PCT/FOREIGN APPLICATION DATA

FOREIGN PRIORITY CLAIMED	COUNTRY CODE	PCT/FOREIGN APPLICATION SERIAL NUMBER	FOREIGN FILING DATE (MONTH DAY YEAR)