

# ACE9040

## AUDIO PROCESSOR for AMPS and TACS CELLULAR PHONES

ACE9040 provides all the speech signal processing and data/SAT tone filtering needed for AMPS or TACS analog cellular telephones.

Transmit voice channel functions comprise a microphone amplifier, soft limiter, bandpass filter, compressor, hard limiter, lowpass filter and a gain controlled amplifier to set deviation level. Additional transmit circuits include a DTMF generator, data and SAT filters, deviation setting amplifiers for data/ST and SAT and a modulation combiner.

ACE9040's receive path comprises a bandpass filter, expander, volume control and power amplifiers to directly drive an earpiece or handsfree transducer.

Gain settings, mute switches and filter characteristics are programmed via a three wire serial interface.

To implement a handsfree function, both transmit and receive paths have rectifiers for signal amplitude monitoring via an external pin and signal path attenuators controlled via the serial interface.

ACE9040 combines minimum power consumption with low external component count. Standby modes greatly reduce supply current and extend battery charge intervals.

### FEATURES

- Low Power and Low Voltage (3-6 to 5-0 V) Operation
- Power Down Modes
- Direct Connections to Microphone and Earpiece
- Compressor with wide operating range:  
Compressor 74 db typ., Expander 36 dB typ.
- SAT Bandpass and Data Lowpass Filters
- Handsfree Operation Supported
- DTMF Generator
- Serial Bus Controlled Gains and Filter Responses
- Part of the ACE Integrated Cellular Phone Chipset
- NMT version available
- TQFP 64 Pin 10x10 mm or 7x7 mm Packages

### APPLICATIONS

- AMPS and TACS Cellular Telephones
- Two-Way Radio Systems

### RELATED PRODUCTS

ACE9040 is part of the following chipset:

- ACE9010 R.F. Front End with VCO
- ACE9020 Receiver and Transmitter Interface
- ACE9030 Radio Interface and Twin Synthesiser
- ACE9050 System Controller and Data Modem

### ABSOLUTE MAXIMUM RATINGS

Supply voltage	6 V
Storage temperature	- 55 °C to + 150 °C
Operating temperature	- 40 °C to + 85 °C
Voltage at any pin	- 0.3 V to V <sub>DD</sub> + 0.3 V

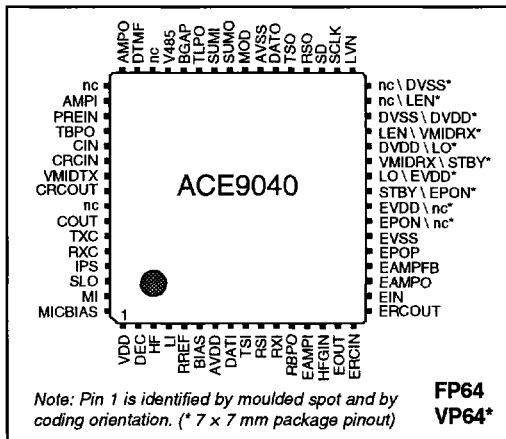


Fig.1 Pin connections - top view

### ORDERING INFORMATION

Industrial temperature range  
TQFP 64 lead 10 x 10 mm, 0.5 mm pitch (FP64)  
**ACE9040/IG/FP1R** - devices shipped in trays  
**ACE9040/PR/FP1R** - pre-production  
TQFP 64 lead 7 x 7 mm, 0.4 mm pitch (VP64)  
**ACE9040/IG/FP2R** - devices shipped in trays  
**ACE9040/PR/FP2R** - pre-production

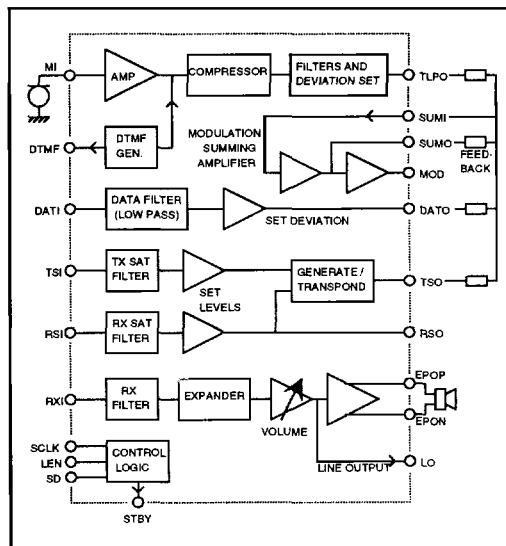


Fig. 2 ACE9040 Simplified Block Diagram

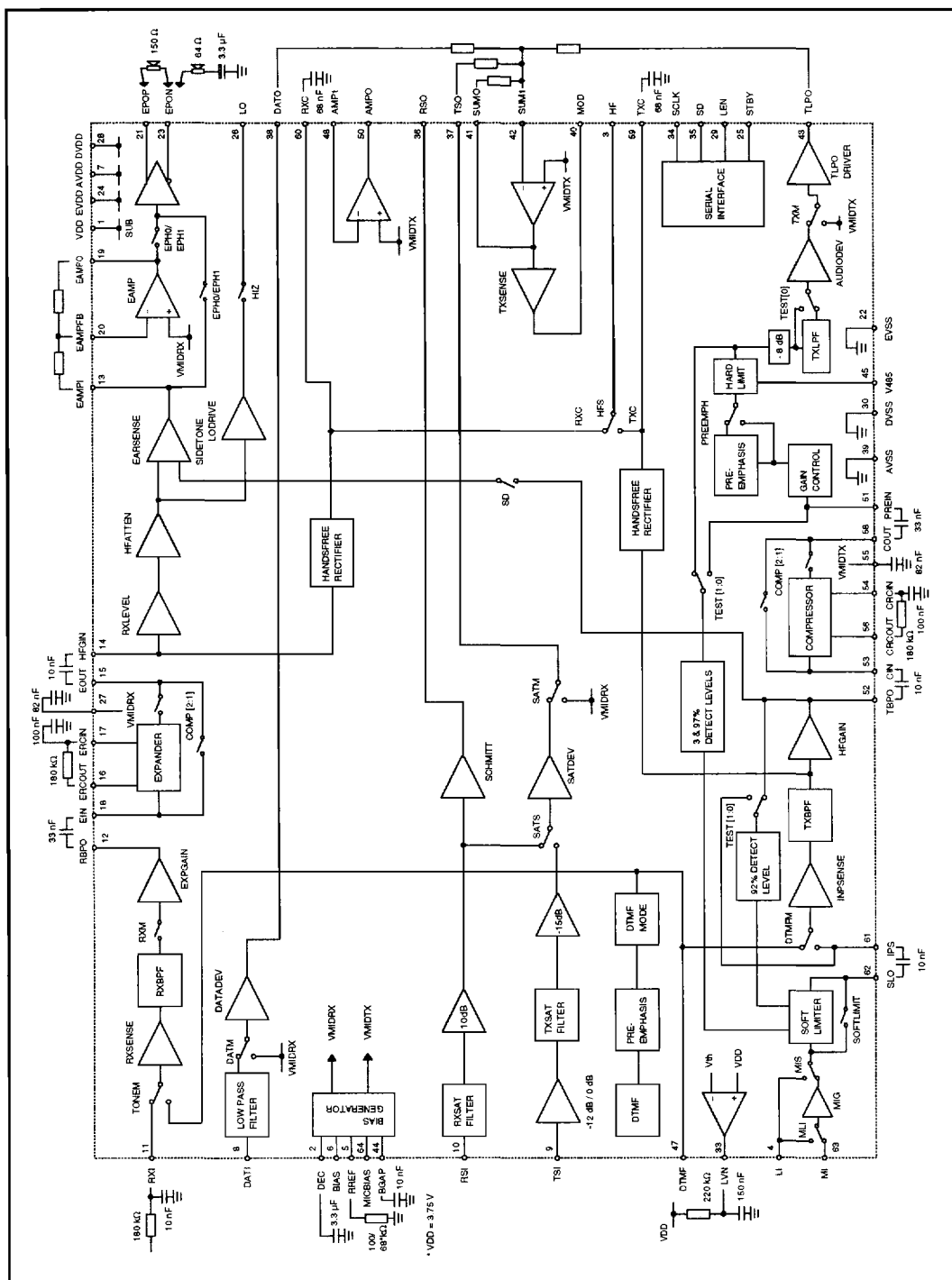


Fig. 3 ACE9040 Detailed Block Diagram

## PIN DESCRIPTIONS

Pin No. FP package	Pin No. VP package	Name	Description
1	1	V <sub>DD</sub>	V <sub>DD</sub> supply to substrate, pin should be at highest d.c. voltage
2	2	DEC	Mid-supply reference decoupling connection, 3.3 $\mu$ F to GND
3	3	HF	Output from TX or RX handsfree rectifier, switched by bit "HFS"
4	4	LI	Line input
5	5	RREF	Reference bias current set for all op-amps by resistor to GND
6	6	BIAS	Buffered mid-supply reference output
7	7	AV <sub>DD</sub>	Analogue V <sub>DD</sub> input
8	8	DATI	Transmit data input
9	9	TSI	SAT path input for locally generated tone
10	10	RSI	SAT path receiver input for received tone
11	11	RXI	Speech path receiver input
12	12	RBPO	Audio output from EXPGAIN block
13	13	EAMPI	Output from EARSENSE amp
14	14	HFGIN	Input to RX volume control and handsfree attenuator
15	15	EOUT	Expander speech output
16	16	ERCIN	Expander time constant input, 180 k $\Omega$ to ERCOUT, 100 nF to GND
17	17	ERCOUT	Expander time constant output, 180 k $\Omega$ to ERCIN
18	18	EIN	Expander speech input, 33 nF to RBPO
19	19	EAMPO	Output from EAMP op-amp
20	20	EAMPFB	Inverting input to EAMP op-amp
21	21	EPOP	Earpiece driver positive output
22	22	EV <sub>SS</sub>	Earpiece V <sub>SS</sub> (GND) supply connection
-	23/24	nc	No connection
23	25	EPON	Earpiece driver negative output
24	26	EV <sub>DD</sub>	Earpiece V <sub>DD</sub> supply input
25	27	STBY	Standby output: low indicates standby state, high is V <sub>DD</sub> output @ 10 mA
26	28	LO	Line output
27	29	VMIDRX	RX path mid-supply reference voltage, 82 nF to GND
28	30	DV <sub>DD</sub>	Digital V <sub>DD</sub>
29	31	LEN	Serial interface latch signal input, rising edge triggered
30	32	DV <sub>SS</sub>	Digital V <sub>SS</sub> (GND) connection
31/32	-	nc	No connection
33	33	LVN	Low supply V <sub>DD</sub> voltage indicator comparator output, reset active low output
34	34	SCLK	Serial interface system clock input
35	35	SD	Serial interface data input
36	36	RSO	Received (regenerated) SAT output
37	37	TSO	Transmit SAT output, regenerated or returned
38	38	DATO	Transmit data filter output.
39	39	AV <sub>SS</sub>	Analogue V <sub>SS</sub> (GND)
40	40	MOD	Modulation output: sum of Speech, Data and SAT
41	41	SUMO	Modulation summing amplifier output
42	42	SUMI	Modulation summing amplifier input
43	43	TLPO	Transmit audio lowpass filter output
44	44	BGAP	Bandgap voltage output, 10 nF to GND
45	45	V485	Hard limiter gain selection for 3.75 V or 4.85 V nominal supplies
46	46	nc	No connection
47	47	DTMF	DTMF tone output
48	48	AMPO	Auxillary op-amp output
49	49	nc	No connection
50	50	AMPI	Auxillary op-amp inverting input (non-inverting internally connected to VMIDTX)
51	51	PREIN	Transmit pre-emphasis filter input
52	52	TBPO	Transmit bandpass filter output
53	53	CIN	Compressor audio input, 10 nF from TBPO
54	54	CRCIN	Compressor time constant input, 100 nF to GND, 180 k $\Omega$ to CRCOUT
55	55	VMIDTX	TX path reference voltage, 82 nF to GND
56	56	CRCOUT	Compressor time constant output, 180 k $\Omega$ to CRCIN
57	57	nc	No connection
58	58	COUT	Compressor audio output, 33 nF to PREIN
59	59	TXC	Transmit handsfree audio level sensing rectifier smoothing filter, 68 nF to GND
60	60	RXC	Received handsfree audio level sensing rectifier smoothing filter, 68 nF to GND
61	61	IPS	Transmit audio gain INPSENSE adjustment block input
62	62	SLO	Soft limiter output
63	63	MI	Microphone input
64	64	MICBIAS	Bias for electret or active microphone

## ELECTRICAL CHARACTERISTICS

These characteristics are guaranteed over the following conditions unless otherwise stated (Note 1):

$$T_{AMB} = -40\text{ }^{\circ}\text{C to } +85\text{ }^{\circ}\text{C}, V_{DD} = 3.6\text{ V to } 5.0\text{ V}$$

Characteristic	Symbol	Value			Unit	Conditions
		Min.	Typ.	Max.		
<b>Supply Current and Power Down Modes</b>						
Operating supply current	$I_{DD}$		13		mA	$V_{DD} = 4.85\text{ V}$ $R_{REF} = 100\text{ k}\Omega$
				13	mA	$V_{DD} = 3.6\text{ V}$ $R_{REF} = 68\text{ k}\Omega$
<b>Standby</b>						
Supply Current	$I_{DD(STBY)}$			200	$\mu\text{A}$	
Attenuation of all inputs signals		40			dB	
Wakeup response time				10	ms	
<b>Sleep (Standby with CLK stopped)</b>						
Supply current	$I_{DD(SLEEP)}$			200	$\mu\text{A}$	STBY bit set
Delay between setting STBY bit and stopping clock		20			$\mu\text{s}$	
Delay to starting clock after wakeup		100			$\mu\text{s}$	

Characteristic	Value			Unit	Conditions
	Min.	Typ.	Max.		
<b>Transmission Path</b>					
<b>Microphone Amp, MI to SLO pins</b>					
Input bias		$V_{DD}/2$			Internal 150 k $\Omega$ bias resistor to $V_{DD}/2$
Microphone input gain	31	32	33	dB	MIS = 1, MLI = 0, MIG = 1
Microphone crosstalk (no MI signal)			-40	dB	MIS = 1, MLI = 0, MIG = 1
<b>Line input, LI to SLO pins</b>					
Input bias		$V_{DD}/2$			Internal 100 k $\Omega$ bias resistor
LI input Gain	-0.5		0.5	dB	MIS = 0, MLI = 0, MIG = 0
LI input Gain	21	22	23	dB	MIS = 1, MLI = 1, MIG = 0
LI input crosstalk (no LI signal)			-40	dB	MIS = 1, MLI = 1, MIG = 1
<b>Soft Deviation Limiter, LI to SLO pins</b>					
Nominal gain		0		dB	Limiter not functioning.
Attenuation range	-30		-29	dB	
Attenuation steps	0.27	0.5	0.67	dB	
Distortion			2	%THD	Output at 1 Vrms
Attack level: Hard limiter output: enabled	2		98	% $V_{DD}$	d.c. input on PREIN
disabled	4		96	% $V_{DD}$	TEST[1:0] = 11
Attack level at TBPO pin		92		% $V_{DD}$	d.c. input at IPS
					TEST[1:0] = 11
Attack time		40		$\mu\text{s}$	Per gain step when signal outside threshold
Decay time		1.68		ms	Per gain step
<b>INPSENSE Gain Stage, IPS to TBPO pins</b>					
Nominal gain	7.5	8	8.5	dB	Input = 100 mVrms, INPS[4:0] = 15, THF = 0 dB,
Gain adjustment range	-12		12.8	dB	Relative to nominal gain
Gain step size	0.6	0.8	1	dB	Input = 100 mVrms, INPS[4:0] = 0 to 31, THF = 0 dB

Note 1. 100% production tested at 25 °C but guaranteed over specified temperature range.

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Characteristic	Value			Unit	Conditions
	Min.	Typ.	Max.		
<b>TX Audio Bandpass Filter TXBPF, IPS to TBPO</b>					
Noise			- 72	dBV	
Distortion			1	%THD	1 Vrms output
Frequency response relative to 1040 Hz	- 60.5		- 38.5	dB	f = 60 Hz
	- 25.5		- 10.5	dB	f = 184 Hz INPSENSE = 0 dB
	- 1.0		+ 0.5	dB	f = 430 Hz THF = 0 dB
	- 0.5		+ 0.5	dB	f = 676 Hz, 1040 Hz
	- 0.5		+ 0.5	dB	f = 1410 Hz, 1900 Hz
	- 1.5		- 0.5	dB	f = 3260 Hz
	- 3.0		- 1.5	dB	f = 3500 Hz
	- 10.5		- 5	dB	f = 4120 Hz
	- 20.5		- 15.5	dB	f = 5590 Hz
	- 90.5		- 30.5	dB	f = 9900 Hz
<b>TX Handsfree Gain Stage HFGAIN</b>					
Nominal gain			0	dB	
Gain range	- 52.5	- 49	- 45.5	dB	THF[2:0] = 0 to 7
Gain step size	6.5	7	7.5	dB	
<b>TX Compressor Stage, CIN to COUT pins</b>					
Unity gain level	671	707	742	mVrms	= Vref (Unaffected level)
Input range	0.79		1000	mVrms	
Linearity CIN to COUT: (Deviation from 2:1 I/O relationship)			± 0.5	dB	CIN = Vref + 3 dB to Vref - 59 dB BW = 300 Hz to 3.4 kHz
Attack time		3		ms	12 dB step: - 8 dB to - 20 dB relative to the unity gain (Vref) level
Decay time		13.5		ms	Attack & Decay levels = 1.5 and 0.75 of steady state final value
Distortion			2	%THD	BW = 300 Hz - 3.4 kHz
Frequency response			± 0.2	dB	BW = 300 Hz - 3.4 kHz
<b>TX Pre-emphasis, PREIN to TLPO pins</b>					
Input impedance		75		kΩ	
Internal compressor and BAR:					
Nominal gain TACS	- 7.5	- 7	- 6.5	dB	COMP[2:1] = 10 or 01 at 1 kHz
Nominal gain AMPS	- 11.5	- 11	- 10.5	dB	
External compressor and bypass:					
Nominal gain AMPS & TACS	- 0.5	0	+ 0.5	dB	COMP[2:1] = 11 or 00 at 1 kHz
Frequency response	5.8	6	6.2	dB/ Octave	TEST[1:0] = 01 PREEMPH = 0 (active) f = 300 to 3400 Hz

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Characteristic	Value			Unit	Conditions
	Min.	Typ.	Max.		
<b>Hard Deviation Limiter</b>					
Gain	13	13.5	14	dB	V485 pin = 0 V (for V <sub>DD</sub> = 3.75 V)
	15.5	16	16.5	dB	V485 pin = V <sub>DD</sub> (for V <sub>DD</sub> = 4.85 V)
<b>Low Pass Filter TXLPF, PREIN to TLPO pins</b>					
Distortion			1.5	%THD	1.5 V <sub>p-p</sub> Output level
Noise			-60	dBV	BW = 30 Hz to 30 kHz
Frequency response TACS/AMPS Relative to 1 kHz	0.3		0.7	dB	f = 307 Hz
	-0.5		+0.5	dB	f = 676 Hz, 1040 Hz
	-0.5		+0.5	dB	f = 1410 Hz, 1900 Hz
	-3.0		-1.5	dB	f = 3010 Hz
	-38.0		-13.0	dB	f = 3500 Hz
	-60.0		-38.0	dB	f = 4120 Hz
	-60.0		-38.0	dB	f = 5590 Hz
	-60.0		-38.0	dB	f = 9900 Hz
	-60.0		-35.0	dB	f = 11870 Hz
	-60.0		-35.0	dB	f = 14950 Hz
Gain	-8.5	-8	-7.5	dB	
Attenuation with TXLPF bypassed		8		dB	TEST[0] = 1
<b>Gain Stage AUDIODEV, PREIN to TLPO</b>					
Nominal gain		0		dB	Output at TLPO
Gain adjustment	-2.8		3.2	dB	
Attenuation step size	0.2	0.4	0.6	dB	Control bits: AUDEV[3:0]
TXM switch attenuation			-60	dB	AUDIODEV = 0 dB
<b>Combined TX path</b>					
Gain with 7 mV at Microphone input	31.0		35.0	dB	INSENSE = 0 dB, V485 = 0 V
Distortion			1.3	% THD	THF & AUDIODEV = 0 dB
Noise			-50.0	dBV	M1 gain = 22 dB, V <sub>DD</sub> = 3.6 V
Output d.c. level	1.65		1.9	V	COMP[2:1] = 00, Softlimit = off

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Characteristic	Value			Unit	Conditions
	Min.	Typ.	Max.		
<b>Receive Path</b>					
<b>RX Input stage RXSENSE, RXI to RBPO pins</b>					
Input bias		$V_{DD}/2$			Internally biased to $V_{DD}/2$ by 150 k $\Omega$
Nominal gain	7.5	8	8.5		
Gain adjustment range	-6		6.4	dB	Input 40 mVrms at RXI
Gain adjustment step size	0.2	0.4	0.6	dB	RXM = 1, RXSENSE = 0 to 31
<b>RX Audio Bandpass Filter RXBPF</b>					
Gain		0		dB	
Distortion			1	%THD	At 1 kHz with 40 mVrms input
Noise			-60	dB	BW= 30 Hz to 30 kHz
Frequency Response	-90.0		-45.0	dB	f = 60 Hz RXSENSE = 0 dB
Relative to 1040 Hz	-15.0		-7.0	dB	f = 184 Hz COMP[1:0] = 00
	-1.0		+0.5	dB	f = 430 Hz
	-0.5		+0.5	dB	f = 676 Hz, 1040 Hz
	-0.5		+0.5	dB	f = 1410 Hz, 1900 Hz
	-3.0		-0.5	dB	f = 3260 Hz
	-10.0		-6.0	dB	f = 3500 Hz
	-40.0		-25.0	dB	f = 4120 Hz
	-40.0		-25.0	dB	f = 5590 Hz
	-90.0		-40.0	dB	f = 9900 Hz
RXM mute switch attenuation			-40	dB	Output switched to $V_{DD}/2$ when muted.
<b>Rx Internal Expander</b>					
Gain EXPAIN: internal	11.8	12.3	12.8	dB	COMP[2:1] = 10
external and bypass		0		dB	COMP[2:1] = 00 or 11
Unity gain level	900	1000	1100	mVrms	= Vref (Unaffected level)
Linearity EIN to EOUT (Deviation from 2:1 input/output)			$\pm 1$	dB	EIN = Vref to Vref - 33 dB BW = 300 to 3400 Hz
Distortion			2	%THD	1 Vrms
Frequency response			$\pm 0.2$	dB	300 to 3400 Hz
Attack time		3		ms	6 dB step (-4 dB and -10 dB) relative to the unity gain (Vref) level at 1 kHz to pin EIN. Output at EOUT.
Decay time		13.5		ms	Attack and delay time levels = 0.57 and 1.5 of final steady state value.
<b>RX Volume Control RXLEVEL</b>					
Nominal Gain	-11	-12	-13	dB	RXV[2:0] = 3, RHF = 0 dB
Gain adjustment range	-9		12	dB	RXV[2:0] = 0 to 7
Gain adjust step size	2	3	4	dB	EARSENSE = 0 dB, Sidetone = 0 dB
<b>RX Handsfree Gain Stage HFATTEN</b>					
HFGAIN to EAMP nominal gain	-0.4	0	0.4	dB	RHF, RVX, EARSENSE = 0 dB, SD = 0
Gain range	-52.5	-49	-45.5	dB	RHF[2:0] = 0 to 7
Gain control step size	6.5	7	7.5	dB	EARSENSE = 0 dB, Sidetone = 0 dB
<b>RX Line Driver LODRIVE</b>					
LO gain	4	5	6	dB	RVX = 0 dB, RHF = 0 dB, HIZ = 1
Distortion			2	%THD	1.0 Vrms output
LO mute	-40			dB	HIZ = 0
Noise			-80	dBV	BW = 30 Hz to 30 kHz
LOoutput during mute	1.5	$V_{DD}/2$	2.1	V	$V_{DD} = 3.6\text{ V}$

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Characteristic	Value			Unit	Conditions
	Min.	Typ.	Max.		
<b>Rx Earpiece Gain Adjustment EARSENSE</b>					
Nominal gain		0		dB	
Total gain	-2.8		3.2	dB	EARS[3:0] = 0 to 15
Gain adjustment step size	0.2	0.4	0.6	dB	RVX = 0 dB, RHF = 0 dB, SD = 0
Distortion			1	% THD	Output 1 Vrms
<b>Rx Sidetone Path, IPS to EAMPI</b>					
Attenuation at EARSENSE amp input	18	19	20	dB	EARSENSE = 0 dB
Sidetone mute			-40	dB	SD = 0
<b>Rx Earpiece Drivers EPOP &amp; EPON</b>					
EPON single ended gain	5.5	6	6.5	dB	Output = 2 Vpp, 120 $\Omega$ EPON to EPOP
EPON single ended d.c. level	1.7	$V_{DD}/2$	1.9	V	$V_{DD} = 3.6\text{ V}$ , EPH1 = 1, EPH0 = 0
EPON single ended a.c. level			-20	dB	
EPON single ended distortion			1	% THD	Output = 2 Vpp
EPON & EPOP differential gain	11.5	12	12.5	dB	150 $\Omega$ ( $\pm 20\%$ ) EPON to EPOP
EPON differential distortion			1	%THD	Output = 4 Vpp, EPH1 = 1, EPH0 = 1
Earpiece mute switch attenuation	40			dB	EPH0 = 0 & EHP1 = 0
EPON external mode: EPON gain relative to EAMPI	5.5		6.5	dB	64 $\Omega$ ( $\pm 20\%$ ) + 3.3 $\mu\text{F}$ to GND, EAMPFB open, Input = HFGIN
EPON distortion			1	% THD	Output = 1.1 Vpp
EPOP output current external mode	-10		10	$\mu\text{A}$	At $V_{DD}$ & 0 V
EPON & EPOP mute			-40	dB	EPH1 = 0, EPH0 = 0
EPOP Noise			-80	dBV	EPH1 = 1, EPH0 = 1
<b>Transmit Data Path</b>					
<b>TX Data Filter 16 kHz &amp; 20 kHz</b>					
Input bias at DATI		$V_{DD}/2$		V	Internally tied via 800 k $\Omega$ resistor to $V_{DD}/2$ .
Nominal gain	-6.5	-6	-5.5	dB	Input = 1 Vrms
Distortion			1.5	%THD	Output = 1.5 Vp-p
Noise			-60	dBV	BW = 30 Hz to 30 kHz
DATM mute switch attenuation	40			dB	
Data Filter frequency response 16 kHz (TACS)	-0.3		0.3	dB	f = 676 Hz, 1040 Hz
	-0.3		0.3	dB	f = 4120 Hz, 12120 Hz
	-1.0		0.0	dB	f = 13960 Hz
	-2.0		-1.0	dB	f = 14950 Hz    DATM = 1
	-3.0		-2.0	dB	f = 16050 Hz    DATADEV = 0 dB
	-3.5		-2.5	dB	f = 16420 Hz    Relative to 1040 Hz
	-4.0		-3.0	dB	f = 17040 Hz    DATAF[1:0] = 10
	-6.0		-4.0	dB	f = 18020 Hz
	-9.5		-7.5	dB	f = 19990 Hz
	-11.0		-9.0	dB	f = 20970 Hz



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Characteristic	Value			Unit	Conditions
	Min.	Typ.	Max.		
<b>TX Data Filter 16 kHz &amp; 20 kHz (continued)</b>					
Data Filter frequency response 20 kHz (AMPS)	-0.3		0.3	dB	f = 676 Hz, 1040 Hz
	-0.3		0.3	dB	f = 4120 Hz, 16050 Hz
	-1.0		0.0	dB	f = 18020 Hz
	-2.0		-0.5	dB	f = 18880 Hz DATAF[1:0] = 11
	-3.0		-1.5	dB	f = 20240 Hz DATM = 1
	-3.5		-2.0	dB	f = 20540 Hz DATADEV = 0dB
	-4.0		-2.5	dB	f = 20970 Hz Relative to 1040 Hz
	-5.0		-3.0	dB	f = 21960 Hz
	-6.0		-4.0	dB	f = 22820 Hz
-8.0		-5.0	dB	f = 24050 Hz	
<b>TX Data Gain Stage DATADEV, DATI to DATO pins</b>					
Nominal gain		0		dB	
Gain adjustment	-2.8		3.2	dB	DATD[3:0] = 7
Gain adjustment steps	0.2	0.4	0.6	dB	DATM = 1 DATAF[1:0] = 00
<b>TXSAT and RXSAT Bandpass Filters 6 kHz</b>					
<b>RXSAT Filter</b>					
RXSAT gain	9		11	dB	Input = 400 mVrms at 6030 Hz, SATDEV = 0 dB, SATM = 1, SATS = 0dB
RXSAT 6 kHz frequency response	-90.0		-35.0	dB	f = 2520 Hz Relative to 6030 Hz
	-90.0		-35.0	dB	f = 3500 Hz SATS = 0 dB,
	-90.0		-35.0	dB	f = 4120 Hz TACS = 1,
	-29.0		-24.0	dB	f = 4980 Hz SATD = 15 (0 dB),
	-0.30		0.50	dB	f = 5900 Hz SATM = 1
	-0.10		0.35	dB	f = 5960 Hz
	-0.10		0.35	dB	f = 6030 Hz
	-0.30		0.50	dB	f = 6090 Hz
	-26.5		-22.0	dB	f = 7010 Hz
	-90.0		-29.0	dB	f = 8060 Hz
	-90.0		-35.0	dB	f = 9040 Hz
	-90.0		-35.0	dB	f = 9290 Hz
RSO Schmitt output	5.5	6.0	6.5	kHz	SATS = 0 dB, TACS = 1

## ELECTRICAL CHARACTERISTICS

These characteristics are guaranteed over the following conditions unless otherwise stated (Note 1):

$$T_{AMB} = -40\text{ }^{\circ}\text{C to } +85\text{ }^{\circ}\text{C}, V_{DD} = 3.6\text{ V to } 5.0\text{ V}$$

Characteristic	Value			Unit	Conditions
	Min.	Typ.	Max.		
<b>TXSAT Filter 6 kHz</b>					
TXSAT gain	-28.0		-26.0	dB	TACS = 1, SATS = 1, Output $\approx$ 1 Vpp at 6030 Hz
TXSAT 6 kHz frequency response	-90.0		-35.0	dB	f = 2520 Hz Relative to 6030 Hz
	-90.0		-35.0	dB	f = 3500 Hz SATS = 1
	-90.0		-35.0	dB	f = 4120 Hz TACS = 1
	-29.0		-24.0	dB	f = 4980 Hz SATD = 15 (0 dB)
	-0.30		0.50	dB	f = 5900 Hz SATM = 1
	-0.10		0.35	dB	f = 5960 Hz
	-0.10		0.35	dB	f = 6030 Hz
	-0.30		0.50	dB	f = 6090 Hz
	-25.0		-22.0	dB	f = 7010 Hz
	-60.0		-35.0	dB	f = 8060 Hz
	-60.0		-35.0	dB	f = 9040 Hz
-80.0		-35.0	dB	f = 9290 Hz	
<b>Gain Stage SATDEV</b>					
Nominal gain		0		dB	
Gain adjustment range	-4.5		4.8	dB	SATS = 0, TACS = 1, SATM = 1
Gain adjustment size	0.1	0.3	0.5	dB	SATD[4:0] = 0 to 31
Distortion			2	% THD	Output = 1.1 Vpp
SATM switch mute attenuation	40			dB	SATM = 0
Noise			-60	dBV	SATD[4:0] = 15 (0 dB)
<b>DTMF Generator</b>					
DTMF output level	60		85	mVrms	$V_{DD} = 3.6\text{ V}$ , high & low tone
DTMF single tone distortion:			5	% THD	Low group, TONEM = 1
			-40	dB	TONEM = 0
			5	% THD	High group, DTMFM = 1
			-40	dB	DTMFM = 0
DTMF high group pre-emphasis	-0.5		0.5	dB	DTWIST = 0
	1.5		2.5	dB	DTWIST = 1
<b>Gain Stage TXSENSE, SUM1 to MOD pins</b>					
Gain	-0.5	0	0.5	dB	Output at MOD.
Gain adjustment	-2.8		3.2	dB	TXSEN[3:0] = 0 to 15
Gain adjustment step size	0.2	0.4	0.6	dB	
Distortion			2	%THD	Output = 1 Vrms
Noise			-80	dBV	BW = 30 Hz to 30 kHz

**ELECTRICAL CHARACTERISTICS**

These characteristics are guaranteed over the following conditions unless otherwise stated (Note 1):

$T_{AMB} = -40\text{ }^{\circ}\text{C}$  to  $+85\text{ }^{\circ}\text{C}$ ,  $V_{DD} = 3.6\text{ V}$  to  $5.0\text{ V}$

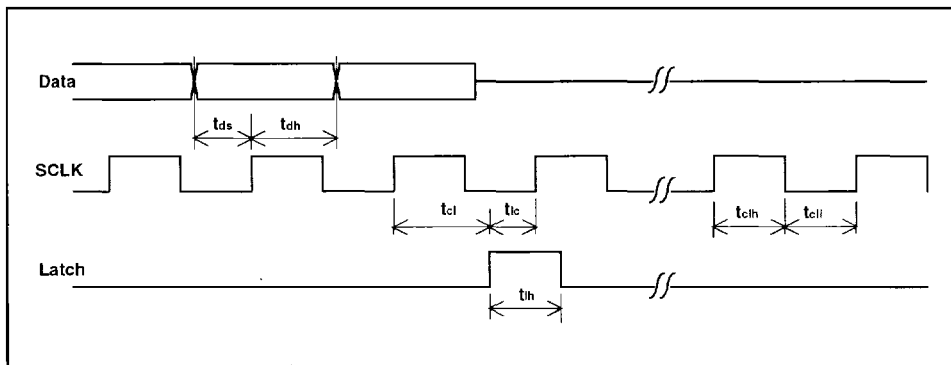
Characteristic	Value			Unit	Conditions
	Min.	Typ.	Max.		
<b>Handsfree Rectifiers</b>					
Attack time		1		ms	Capacitors to GND = 68 nF at TXC & RXC pins
Decay time		35		ms	
TX nominal output levels:					Input at IPS = 0 mVrms Input at IPS = 40 mVrms, 1 kHz INPSENSE = 0 dB, HFS = 0 HFP = 1
d.c. offset	0.3		0.5	V	
a.c. level	1.4	1.5	1.6	V	
RX nominal output levels:					Input at HFGIN = 0 mVrms Input at HFGIN = 100 mVrms, 1 kHz HFS = 1, HFP = 1
d.c. offset	0.3		0.5	V	
a.c. level	1	1.1	1.2	V	
<b>Voltage Outputs</b>					
DEC & BIAS output voltage	1.70	$(V_{DD}-V_{SS})/2$	1.9	V	$V_{DD} = 3.6\text{ V}$ , 3.3 $\mu\text{F}$ decoupling to GND at DEC pin
Microphone Bias:      Source 1 mA Sink 1 mA Microphone disabled	2.7		2.9	V	$V_{DD} = 3.6\text{ V}$ MLI = 0, MIS = 1, PD = 0 MLI = 0, MIS = 1 MLI = 0, MIS = 0
	2.7		2.9	V	
	-0.1		0.10	V	
External Reference Resistor connected from pin RREF to GND		100		k $\Omega$	$V_{DD} = 4.85\text{ V}$
		68		k $\Omega$	$V_{DD} = 3.75\text{ V}$
Bandgap at BGAP	1.10		1.35	V	
<b>LVN Supply Voltage Comparator</b>					
$V_{DD} = 3.6\text{ V}$	-10		10	$\mu\text{A}$	Default condition (high at $V_{DD} - 0.2\text{ V}$ ) (low at $0.4\text{ V}$ )
$V_{DD} = 3.1\text{ V}$	2.00		20.0	mA	
LVN comparator operating lower limit of $V_{DD}$		1		V	

**ELECTRICAL CHARACTERISTICS**

These characteristics are guaranteed over the following conditions unless otherwise stated (Note 1):

$T_{AMB} = -40\text{ }^{\circ}\text{C}$  to  $+85\text{ }^{\circ}\text{C}$ ,  $V_{DD} = 3.6\text{ V}$  to  $5.0\text{ V}$

Characteristic	Symbol	Value			Unit	Conditions
		Min.	Typ.	Max.		
<b>DC Characteristics</b>						
Logic input high	$V_{IH}$	$0.7 \times V_{DD}$		$V_{DD} + 0.3$	V	
Logic input low	$V_{IL}$	-0.3		$0.3 \times V_{DD}$	V	
Input Capacitance	$C_{in}$		10		pF	
logic inputs leakage current	$I_{ILK}$			$\pm 1$	$\mu\text{A}$	SCLK, SD, LEN inputs $V_{DD} = 3.6\text{ V}$ & $0.0\text{ V}$
<b>AC Characteristics</b>						
Clock input frequency: Serial interface & SC Filters	$f_{CLK}$		1.008		MHz	Deviation from 1.008 MHz
		-100		100	ppm	
Clock duty cycle	D	40	50	60	%	
Number of clock rising edges to input data		24			clock cycles	
Clock cycles between latch pulses on LEN		30			clock cycles	
Clock cycles before power-up and after powerdown		8			clock cycles	
Data setup time	$t_{ds}$	80			ns	$V_{DD} = 3.75\text{ V} \pm 0.1\text{ V}$ .
Data hold time	$t_{dh}$	80			ns	$T_A = -40\text{ }^{\circ}\text{C}$ to $+85\text{ }^{\circ}\text{C}$
Clock low	$t_{dL}$	400		600	ns	
Clock high	$t_{dH}$	400		600	ns	
Clock high to latch high	$t_{cl}$	440			ns	
Latch high to clock high	$t_{lc}$	220			ns	
Latch high	$t_{lh}$	240			ns	
Rise and fall times				50	ns	All digital inputs
Power Supply Rejection Ratio:	PSRR					$V_{DD} = 3.8\text{ V} + 100\text{ mVpp}$ (a.c.) COMP[2:1] = 00.
TX path (LI to TLPO)				-20	dB	LI = 0 dB, V485 = 13.5 dB INPSENSE = 0 dB, softlimit on.
RX path (RXI to EPOP)				-30	dB	AUDIODEV, RXSENSE = 0 dB THF, RHF, EARSENSE = 0 dB RXV, SIDETONE off.
Crosstalk between RX and TX path				TBA	dB	



Serial Interface Input Timing

TYPICAL FREQUENCY RESPONSES

**TXBPF Relative Response vs Frequency**

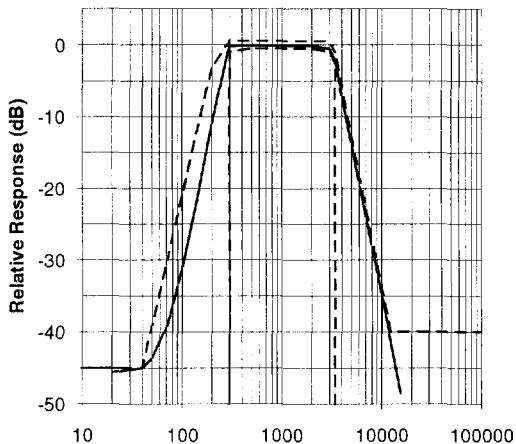


Figure 4

**Pre-emphasis Relative Response vs Frequency**

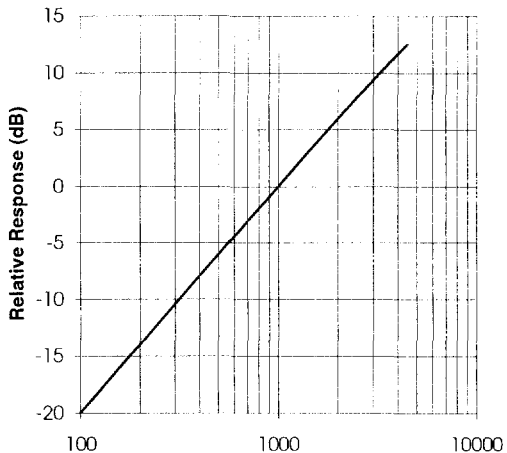


Figure 5

**TXLPF Relative Response vs Frequency**

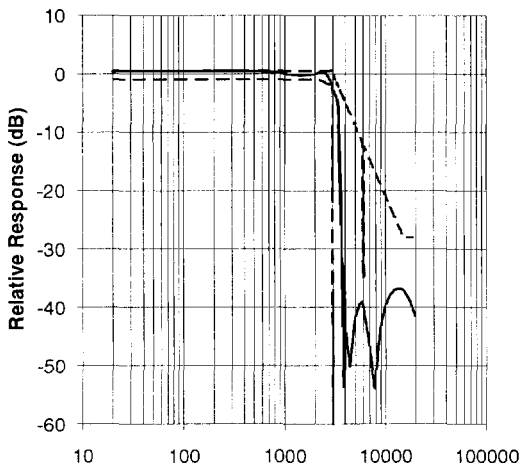


Figure 6

**RXBPF Relative Response vs Frequency**

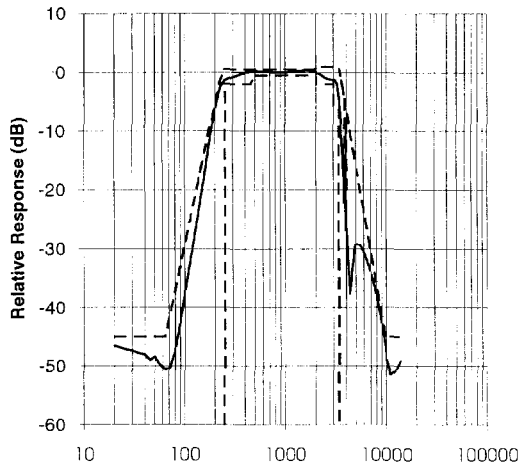


Figure 7

TYPICAL FREQUENCY RESPONSES

Transmit Overall Relative Response vs Frequency

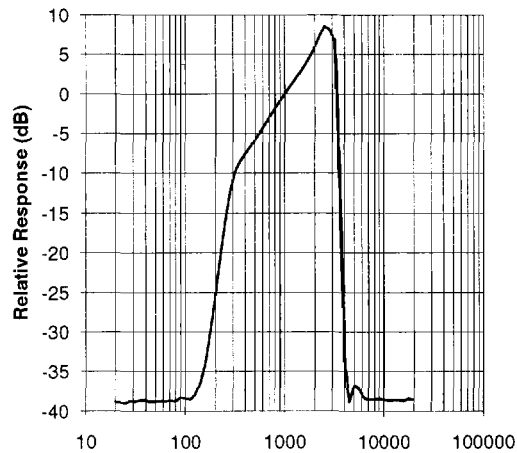


Figure 8

SAT Filter Relative Response vs Frequency

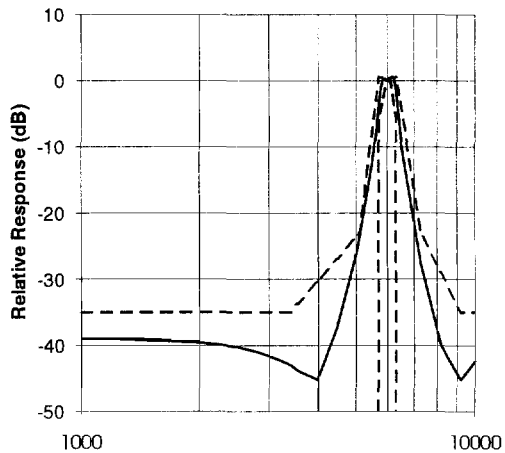


Figure 9

## DESCRIPTION

ACE9040 combines all the voice, data and signalling processing circuits for analog cellular telephones operating with the AMPS or TACS systems.

Transmit channel functions comprise a microphone amplifier, soft limiter, bandpass speech filter, compressor, pre-emphasis filter, hard limiter, lowpass transmit filter and a gain control stage to set the deviation. Additional transmit circuits include a DTMF generator, a lowpass filter for either control data or signalling tone (ST), filters for supervisory audio tone (SAT), either transponded or locally re-generated, and deviation setting amplifiers for Data, ST and SAT. The outputs from the transmit functions feed a modulation combiner whose gain can be adjusted before driving a modulator and external power amplifier.

ACE9040's receive path consists of a bandpass filter, expander, volume control and power amplifier to directly drive the earpiece, either differentially or in single ended modes. Sidetone and DTMF tones can be introduced into the receive path.

Gain settings, filter characteristics and system control is programmed via a three wire serial interface to give optimum operation with either the AMPS or TACS analog cellular systems.

To implement a handsfree function, both transmit and receive paths have rectifiers which enable signal amplitude monitoring via an external pin and signal path attenuators controlled via the serial interface.

All filter characteristics are set by ratioed on-chip components and by a fixed externally input clock rate of 1-008 MHz and do not need trimming, filter response options are selected via the serial interface.

Gain adjustments for different system specifications and component tolerancing are set via the serial interface using gain control blocks in the transmit and receive signal paths. These eliminate the need for any mechanically adjusted

potentiometers. Some gain levels change automatically when the control bits for one of the standards are set, others are under user control.

Power saving operates when an individual block is deselected and for the whole circuit when in Standby. The circuit combines high performance with minimum power consumption and uses as few external components as possible.

## SERIAL CONTROL BUS

All functions are controlled via a three wire serial interface. Input is via pins SD for serial data, SCLK for the clock input and LEN for the control message latch signal.

Incoming data bits are clocked in on the rising edges of SCLK clock input. At the end of each control message comprising three 8-bit data bytes, the rising edge of the LEN pulse latches in the data. A system controller should clock data out on clock falling edges to ensure the maximum timing margins.

The SCLK clock input must be at 1-008 MHz and continuous whenever the ACE9040 is active because ACE9040's switched capacitor filters use clocks derived from SCLK to set frequency responses.

ACE9040 expects a minimum of 30 clock cycles between LEN latch pulses, including the clock cycle containing the latch pulse. A minimum of 8 clock cycles before the beginning of an Operate command or after a Standby command are expected.

Three data bytes DATA1, DATA2, and DATA3 contain bits for system selection, control and mute switches, gain control and filter response settings, as shown in figure 10. The last two bits of DATA3, DATA3[1] and [0], determine the message type, either "Operation", "Initializing mode 0", "Initializing mode 1" or "Handsfree". The details of these four modes are described in tables 2 to 19.

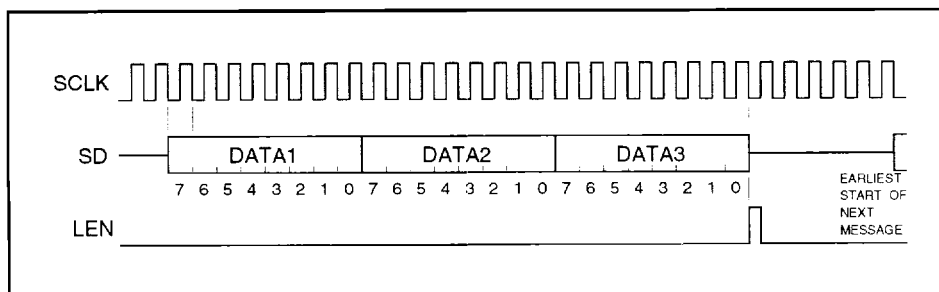


Fig. 10 Serial Receive Bus Timing

DATA3[1]	DATA3[0]	Mode
0	1	"Operation"
0	0	"Initializing mode 0"
1	0	"Initializing mode 1"
1	1	"Handsfree"

Table 1 Mode Selections

CONTROL BUS: OPERATION MODE

Data bit	Bit Name	Function	Effect when at 0	Effect when at 1
DATA3:				
[0]	DATA3[0]	Bus Mode Select	Must be "1"	Must be "1"
[1]	DATA3[1]	Bus Mode Select	Must be "0"	Must be "0"
[2]	PD	Power Down	Operate	Standby
[5:3]	RXV[2:0]	Receive Volume	See table 3	See table 3
[6]	HIZ	Line Output Drive Enable	Off	On
[7]	EPH1	Earpiece Mute	See table 5	See table 5
DATA2:				
[0]	RXM	Receiver Audio Mute	Muted	On
[1]	SATM	Transmit SAT Mute	Muted	On
[2]	DATM	Transmit Data Mute	Muted	On
[3]	TXM	Transmit Audio Mute	Muted	On
[4]	MLI	Microphone Select	See table 4	See table 4
[5]	DTMFEN	Enables each DTMF tone selection	Disabled	Enabled
[7:6]	DTMFMODE[1:0]	DTMF Tone Select	See table 6	See table 6
DATA1:				
[0]	DTMFM	Transmit DTMF Switch	Speech	DTMF
[1]	TONEM	DTMF RX path Confirm Tone Switch	Speech	DTMF
[5:2]	DTMF[3:0]	DTMF Code Select	See table 7	See table 7
[6]	MIS	Line Input Select	See table 4	See table 4
[7]	PREEMPH	Transmit Pre-emphasis Bypass	Active	Bypassed

Table 2 "Operation Mode", DATA3[1:0] = 01.

RXV[2]	RXV[1]	RXV[0]	Gain in dB
0	0	0	-21
0	0	1	-18
0	1	0	-15
0	1	1	-12
1	0	0	-9
1	0	1	-6
1	1	0	-3
1	1	1	0

Table 3 Receiver Volume Control Nominal Levels set by RXV[2:0].

EPH1	EPH0	Function
0	0	Earphone mute
0	1	External Earpiece (EPON to ground)
1	0	Single ended output (EPON to EPOP)
1	1	Differential output (EPON and EPOP)

Table 5 Earphone Mode Select (EHP0 in Initializing mode 0)

DTMFMODE[1:0] bits 1 and 0	Tone generated
0 and 0	No tone.
0 and 1	Low frequency only.
1 and 0	High frequency only.
1 and 1	Dual tones.

Table 6 DTMF Mode Selection

MLI	MIS	Function
0	0	LI selected, no gain
0	1	MI selected, Micamp gain
1	0	NOT ALLOWED
1	1	LI selected, Micamp gain

Table 4 Microphone Input Select

DTMF[3:0] bits 3 2 1 0	Keypad legend	Low Freq. Hz.	High Freq. Hz.
0 0 0 0	1	697	1209
0 0 0 1	2	697	1336
0 0 1 0	3	697	1477
0 0 1 1	A	697	1633
0 1 0 0	4	770	1209
0 1 0 1	5	770	1336
0 1 1 0	6	770	1477
0 1 1 1	B	770	1633
1 0 0 0	7	852	1209
1 0 0 1	8	852	1336
1 0 1 0	9	852	1477
1 0 1 1	C	852	1633
1 1 0 0	*	941	1209
1 1 0 1	0	941	1336
1 1 1 0	#	941	1477
1 1 1 1	D	941	1633

Table 7 DTMF Tones



## CONTROL BUS: INITIALIZING MODE 0

Data bit	Bit Name	Function	Effect when at 0	Effect when at 1
DATA3:				
[0]	DATA3[0]	Bus Mode Select	Must be "0"	Must be "0"
[1]	DATA3[1]	Bus Mode Select	Must be "0"	Must be "0"
[2]	EPH0	Earpiece Mode Select	See table 5	See table 5
[3]	DTWIST	DTMF Pre-emphasis	0 dB	2 dB
[4]	PDLVC	Power Supply Comparator	Active	Power Down
[6:5]	COMP[2:1]	Compander Control	See table 9	See table 9
[7]	INPS[0]	Transmit Audio Gain Adjust	See table 10	See table 10
DATA2:				
[3:0]	INPS[4:1]	Transmit Audio Gain Adjust	See table 10	See table 10
[7:4]	RXSEN[3:0]	Receive Audio Gain Adjust	See table 11	See table 11
DATA1:				
[0]	RXSEN[4]	Receive Audio Gain Adjust	See table 11	See table 11
[5:1]	SATD[4:0]	SAT Modulation Gain	See table 12	See table 12
[6]	SD	Sidetone Enable	Off	On
[7]	SOFTLIMIT	Enables Softlimiter	Off	On

Table 8 "Initializing Mode 0", DATA3[1:0] = 00.

COMP[2]	COMP[1]	Mode
0	0	Bypass Compander
1	0	Internal Compander
1	1	External Compander
0	1	BAR Signal Input Mode (RBPO at high impedance)

Table 9 Compander Operating Modes set by COMP[2:1].

INPS[4]	INPS[3]	INPS[2]	INPS[1]	INPS[0]	Gain (dB)
0	0	0	0	0	- 12.0
0	0	0	0	1	- 11.2
:	:	:	:	:	:
:	:	:	:	:	:
1	1	1	1	0	+ 12.0
1	1	1	1	1	+ 12.8

Table 10 INPSENSE Transmit Audio Nominal Gain settings by INPS[4:0].

RXSEN[4]	RXSEN[3]	RXSEN[2]	RXSEN[1]	RXSEN[0]	Gain (dB)
0	0	0	0	0	- 6.0
0	0	0	0	1	- 5.6
:	:	:	:	:	:
:	:	:	:	:	:
1	1	1	1	0	+ 6.0
1	1	1	1	1	+ 6.4

Table 11 RXSENSE Receive Audio Nominal Gain settings by RXSEN[4:0].

SATD[4]	SATD[3]	SATD[2]	SATD[1]	SATD[0]	Gain (dB)
0	0	0	0	0	- 4.5
0	0	0	0	1	- 4.2
:	:	:	:	:	:
:	:	:	:	:	:
1	1	1	1	0	+ 4.5
1	1	1	1	1	+ 4.8

Table 12 SATDEV Transmit SAT Nominal Modulation Gain settings by SATD[4:0].

CONTROL BUS: INITIALIZING MODE 1

Data bit	Bit Name	Function	Effect when at 0	Effect when at 1
DATA3: [0] [1] [3:2] [4] [6:5] [7]	DATA3[0] DATA3[1] TEST[1:0] SATS DATAF[1:0] TACS	Bus Mode Select Bus Mode Select Test Mode Select SAT source Select Data Filter Bandwidth Select Cellular System Select	Must be "0" Must be "1" see table 14 see table 15 *	Must be "0" Must be "1" see table 14 see table 15 TACS†
DATA2: [3:0] [7:4]	TXSEN[3:0] DATD[3:0]	Combined Modulation Gain Data Modulation Gain	see table 16 see table 16	see table 16 see table 16
DATA1: [3:0] [7:4]	AUDEV[3:0] EARS[3:0]	Audio Modulation Gain Earpiece "EARSENSE" Gain	see table 16 see table 16	see table 16 see table 16

\*See table 19.

Table 13 "Initializing Mode 1", DATA3[1:0] = 10.

TEST[1]	TEST[0]	Test Mode
0	0	Operate mode (not in test mode)
1	0	Test DTMF
0	1	Bypass TXLPF
1	1	Bypass TXLPF & Softlimiter test

Table 14 Test Modes selected by TEST[1:0].

DATAF[1]	DATAF[0]	Cut-off Frequency
1	0	16 kHz
1	1	20 kHz

Table 15 Data Filter Bandwidths selected by DATAF[1:0].

TXSEN[3] DATD[3] AUDEV[3] EARS[3]	TXSEN[2] DATD[2] AUDEV[2] EARS[2]	TXSEN[1] DATD[1] AUDEV[1] EARS[1]	TXSEN[0] DATD[0] AUDEV[0] EARS[0]	Gain (dB)
0	0	0	0	- 2.8
0	0	0	1	- 2.4
:	:	:	:	:
:	:	:	:	:
1	1	1	0	+ 2.8
1	1	1	1	+ 3.2

Table 16 TXSENSE, DATADEV, AUDIODEV, and EARSENSE Nominal Gains set, respectively, by TXSEN[3:0], DATD[3:0], AUDEV[3:0], and EARS[3:0].

## CONTROL BUS: HANDSFREE MODE

Data bit	Bit Name	Function	Effect when at 0	Effect when at 1
DATA3: [0] [1] [4:2] [7:5]	DATA3[0] DATA3[1] THF[2:0] RHF[2:0]	Bus Mode Select Bus Mode Select Transmit Handsfree Gain Receive Handsfree Gain	Must be "1" Must be "1" see table 18 see table 18	Must be "1" Must be "1" see table 18 see table 18
DATA2: [0] [1] [2] [4:3] [5] [6] [7]	HFP HFS MIG not used AMPS not used -	Handsfree Rectifiers Power HF Output Connection Path Microphone Amplifier Gain - Cellular System Select - -	Off Transmit + 22 dB - † - must be "0"	On Receive + 32 dB - AMPS† - must be "0"
DATA1: [2:0] [5:3] [7:6]	- not used not used	- - -	must be "0" - -	must be "0" - -

†See table 19.

Table 17 "Handsfree Mode", DATA3[1:0] = 11.

THF[2] RHF[2]	THF[1] RHF[1]	THF[0] RHF[0]	Gain (dB)
0	0	0	0
0	0	1	-7
0	1	0	-14
0	1	1	-21
1	0	0	-28
1	0	1	-35
1	1	0	-42
1	1	1	-49

Table 18 TXHFGAIN and RXHFGAIN Handsfree Attenuator Nominal Gains, set by THF[2:0] and RHF[2:0].

CIRCUIT OPERATING MODES

ACE9040 has three operating modes: Operate, Standby and Sleep. In Operate mode all parts of the circuit are active, except for any explicitly powered down and the DTMF generator which only powers up when tones are generated. The pin STBY is pulled high in Operate mode to supply  $V_{DD}$  to external audio circuits, such as a compander.

Standby Mode

Standby mode is used when the cellular terminal is waiting for a call and is selected by an "Operation mode" control message with bit PD at "1". In this mode all analog circuits, data paths, filters and their clock drivers are powered down giving a greatly reduced supply current. In standby mode all switch and level controls retain their previous state, the pin STBY is not driven removing the  $V_{CC}$  supply from external circuits. To leave Standby mode an operate command should be given by using an "Operation mode" control message with bit PD at "0".

Sleep Mode

Sleep mode is the same as Standby but without a clock input. To enter sleep mode the standby command should be given and after a delay of at least 20  $\mu$ s the SCLK clock stopped. To wake-up from sleep mode, the SCLK clock must be started and after a delay of at least 100  $\mu$ s an operate command given.

Power on Reset

At power up ACE9040 is put into Standby mode. ACE9040 is set up by the controlling processor via the serial interface using four control messages: "Operation", "Initializing mode 0", "Initializing mode 1" and "Handsfree". Usually the "Operation" control message would be sent last as it contains the power down/up bit PD.

CELLULAR SYSTEM SELECTION

Two control bits are used to set the filter responses and gain levels for the AMPS or TACS cellular systems. These bits are TACS in control message "Initializing mode 1" and AMPS in control message "Handsfree mode" and select the system as shown in table 19.

Bits		System Selected
AMPS	TACS	
0	0	Not valid
0	1	TACS
1	0	AMPS
1	1	Not valid

Table 19 Cellular System Selection

TRANSMIT VOICE PATH

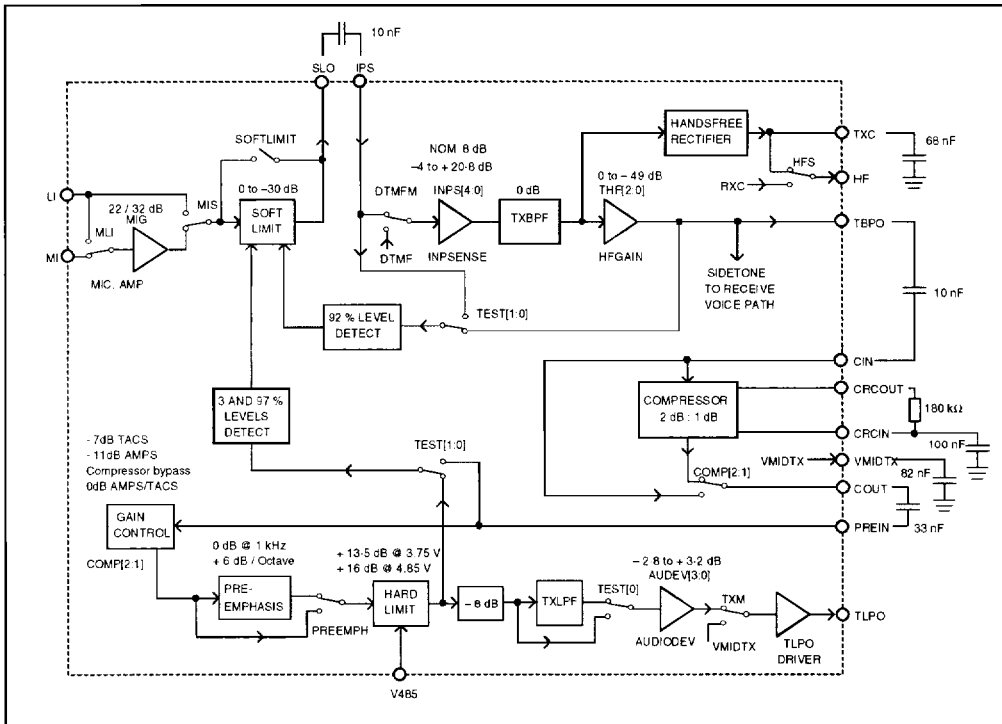


Fig. 11 Transmit Voice Path

## TRANSMIT INPUT SIGNAL PRECONDITIONING

### Microphone Amplifier (MIG)

Microphone signals input at MI via switch MLI are amplified by a gain selectable amplifier of either 22 dB or 32 dB, controlled by the MIG bit in the "Handsfree mode" control message. The microphone amplifier's input can also be connected to the line input LI via the MLI switch. The microphone amplifier's output drives the soft limiter via switch MIS which also allows higher level signals from the line input LI to bypass the microphone amplifier to drive the soft limiter directly. These two switches are controlled by the MIS and MIL bits in the "Operation mode" control message.

### Soft Limiter

Signal amplitude is restricted without clipping to the correct level for F.M. deviation by the soft limiter. This operates as an AGC system, controlled by the signal amplitude at the compressor input and the hard limiter output later in the signal processing path. If the signal is too large at either of these points the soft limiter forward gain is reduced. The nominal soft limiter gain range of 0 to -30 dB in 64 steps of 0.5 dB covers all normal volume changes occurring during a conversation. Soft limiter attack and decay times are set by internal clocks derived from SCLK and ramp the gain steps at nominal rates of one step down every 40  $\mu$ s when the signal is too large and one step up every 1.68 ms when the signal is too small.

The soft limiter output at pin SLO is externally coupled with a capacitor to the input pin IPS of the following gain adjust stage, INPSENSE. When the speech channel is used to send DTMF signalling tones a switch, controlled by bit DTMFM in the "Operation mode" control message, selects the internal DTMF signal rather than the speech signal at input IPS.

For test purposes TEST[1:0] bits in the "Initializing Mode 1" control message can configure switches to give access to the softlimiter comparator inputs. When TEST[1:0] bits are high the 3 & 97 % window comparator is switched from the hard limiter output to PREIN input and the 92 % comparator is switched to IPS input.

### INPSENSE AMPLIFIER and TXBPF FILTER

Both microphone and transmit voice path tolerances are trimmed in the INPSENSE gain adjustment block following the soft limiter and DTMF switch. INPSENSE has a nominal gain of 8.0 dB with a trim range of -12.0 to +12.4 dB relative to the nominal gain set by bits INPS[4:0] in the "Initializing mode" control message.

Transmit signal preconditioning is completed by a bandpass filter TXBPF to limit the audio signal to a speech bandwidth of 300 Hz to 3.4 kHz. This filter uses switched capacitor techniques and is preceded by an anti-alias filter and followed by a smoothing filter to remove the clock from its output. The typical frequency response is shown in figure 4 which also shows the mask defined by type approval limits.

## HANDSFREE FUNCTIONAL BLOCKS

Between the transmit bandpass filter and the compressor two extra functions are included for use with handsfree operation. Firstly an attenuator HFGAIN is provided to give progressive signal reduction in handsfree mode with a range of 0 to -49 dB in 7 dB steps, set by bits THF[2:0] in the "Handsfree mode" control message. The HFGAIN attenuator's output drives pin TBPO which is externally capacitively coupled to the compressor input at pin CIN and internally drives both the soft limiter and sidetone circuits.

The second function provided for handsfree operation is a signal rectifier whose output, filtered by an external capacitor at pin TXC, drives output pin HF via switch HFS with a d.c. transmit level. Switch HFS is controlled by bit HFS in the "Handsfree mode" control message. Under control of bit HFS both receive and transmit levels are available at HF output pin for external comparison to implement the handsfree function. The handsfree system is further described in the section HANDSFREE OPERATION.

## COMPRESSOR

ACE9040 provides a 2:1 compressor to halve the transmit dynamic range as required by analog cellular systems. Within the operating signal range each 2 dB change in input level gives a 1 dB change in output level. A transmit signal is input through pin CIN and output on pin COUT, the signal is referenced to a mid-supply voltage. CRCIN and CRCOUT are connections for the external attack and delay time constant setting components.

The compressor's nominal unity gain (unaffected) level is 707 mV. Above this level the signal at CIN is attenuated and below this level the signal is amplified to achieve the 2:1 dB compression. Table 20 gives the nominal, Vmax and Vmin levels at CIN and nominal levels at COUT corresponding with the TACS and AMPS systems for 0 dB, maximum and minimum deviation.

For the usual attack time of 3.0 ms and decay time of 13.5 ms a 180 k $\Omega$  resistor is connected between CRCIN and CRCOUT pins and a 100 nF capacitor between CRCIN and GND pins. An 82 nF capacitor should be connected between the VMIDTX and GND pins. Attack and decay time is measured with a 12 dB step, -8 dB to -20 dB relative to the unaffected level. Attack and decay times are respectively defined at points on the output envelope where it reaches x 1.5 and x 0.75 of the final steady state level.

External compressor connections allow the use of external coupling capacitors to remove d.c. offsets and optionally an external compander. The compressor can be internally bypassed allowing use of ACE9040 without companding in non-cellular applications, or for test purposes. Bits COMP[2:1] in the "Initializing mode 0" control message control the operation of the internal compander and are used to switch both the transmit compressor and receive expander into or out of the signal path. When not in use the internal compressor and expander are both powered down.

System	Vmax at CIN		Vmin at CIN		Nominal Input at CIN (0dB) levels mVrms	Nominal Output at COUT levels mVrms
	Vrms	dB	mVrms	dB		
TACS	1.0	+25	1.77	-30	56	200
AMPS	1.0	+23	2.25	-30	71	225

Table 20 Compressor CIN and COUT signal levels for TACS and AMPS

**FINAL MODULATION PREPARATION**

**Pre-emphasis**

A pre-emphasis filter follows the compressor to boost the amplitude of higher audio frequencies by tilting the frequency response by 6 dB per octave across the whole speech band as shown in figures 5 and 8. To prevent overload in the pre-emphasis filter the signal first passes through an attenuator set to suit the system in use. If an external compander is used or the companding function is bypassed the gain is set to 0 dB. When using the internal compander the gain is set to -7 dB for TACS or -11.0 dB for AMPS. Compander bypass is determined by control bits COMP[2:1]. The pre-emphasis filter and attenuator input is pin PREIN and the output is an internal connection to the hard limiter. The pre-emphasis filter, but not the attenuator, can be bypassed if the PREEMPH bit in the "Operation mode" control message is set to "1".

**Hard Limiter**

To ensure compliance with the peak deviation specification for cellular telephone systems, a hard limiter follows the pre-emphasis filter to remove any transient level changes that have passed through the soft limiter. This limiter will handle large signals and has symmetrical clipping levels close to the supply rails  $V_{DD}$  &  $V_{SS}$  (GND). To ensure clipping at the same hard limiter input signal level with both the nominal power supply voltages, hard limiter gain is adjusted

by an external pin "V485". For the nominal supply voltages of 4.85 V and 3.75 V gain is respectively 16 dB (V485 pin at "1") and 13.5 dB (V485 pin at "0"). An 8 dB attenuator follows the limiter to prevent any further clipping of the signal in the following transmit lowpass filter.

**TX Lowpass Filter TXLPF**

A TXLPF lowpass filter with an optimised stop band response limits the signal bandwidth to a cut-off frequency of 3.0 kHz, the frequency response is shown in figure 6. The combined frequency response of the pre-emphasis and lowpass filter stages is shown in figure 8. It is possible for test purposes to bypass this lowpass filter by setting bit TEST[0] in the "Initializing mode 1" control message to a "1".

**Speech Deviation Level Setting**

A controlled gain stage AUDIODEV sets the output level to give the required FM deviation for speech. The gain is set by bits AUDEV[3:0] in the "Initializing mode 1" control message. AUDIODEV is followed by a transmit audio mute switch enabled by bit TXM in the "Operation mode" control message. A buffer drives output pin TLPO with the transmit speech signal (and DTMF when in use) which is added with DATA/ST and SAT tones in the modulation combiner.

**TRANSMIT DATA AND DTMF PATHS**

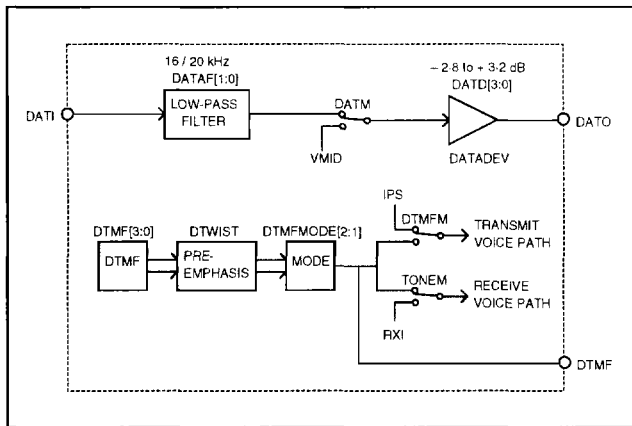


Fig. 12 Transmit Data and DTMF Paths

**Transmit Data**

Data communication from mobile terminals to base stations in the AMPS and TACS cellular phone systems takes place over the Reverse Control Channel (RECC) during call set-up and in short bursts over the Reverse Voice Channel (RVC) during a call.

RECC or RVC data is transmitted for AMPS or TACS as a 10 kHz or 8 kHz Manchester Coded FSK signal respec-

tively. The data signal is generated by the ACE9050 "System Controller and Data Modem" or similar digital circuit to drive ACE9040's DATI input pin. The DATI input data signal is filtered using a 4th order Butterworth lowpass filter with nominal -3 dB points of 16 kHz for TACS, or 20 kHz for AMPS. This filter is implemented using switched capacitor techniques and is preceded by a continuous time anti-alias

filter, the output buffer includes a clock rejection filter. The cut-off frequencies are programmed by bits DATAF[1:0] in the "Initializing mode 1" control message. Filtered data passes through the mute switch DATM and a variable gain stage DATADEV with a range of  $-2.8$  to  $+3.2$  dB to set the required level of deviation. The mute switch is controlled by bit DATM in the "Operation mode" control message. DATADEV is controlled by bits DATD[3:0] in the "Initialising Mode 1" control message. The data signal is buffered out to pin DATO to drive the modulation combiner.

## DTMF

DTMF tones are generated when commanded via the serial interface and conform to the standard CCITT frequencies. All 16 standard tone pairs or any individual tone can be generated. To select DTMF tones data bits for transmission DTMFEN, DTMFMODE[1:0] and DTMF[3:0] need to be set using an "Operation mode" control message. Data bits DTMFMODE[1:0] select low, high or both tones of the pair as shown in table 6. Bits DTMF[3:0] select the tone pair as shown in table 7. DTMFEN set to "1" enables DTMF operation. To change DTMF tones, an "Operation mode" control message

with DTMFEN set to "0" must be sent to cancel the previous selection.

An optional pre-emphasis of 2 dB of the high frequency tone group above the level of the low frequency group is enabled by bit DTWIST in the "Initializing mode 0" control message.

DTMF tones can be selected to replace the speech in either or both the transmit and receive paths. In the transmit path setting bit DTMFM to "1" in the "Operation mode" control message will connect the DTMF signal to INPSENSE gain adjustment block's input in place of the speech signal. In the receive path setting bit TONEM to "1" in the "Operation mode" control message will connect the DTMF signal to the input to RXSENSE gain adjustment block in place of the speech input at pin RXI.

DTMF signals are generated as sinewaves by an internal digital to analog converter and are smoothed by the transmit and receive filters. DTMF waveforms start and stop at a zero crossing to avoid transients in the filters and to limit their bandwidth. The DTMF signal is brought out directly on pin DTMF without further buffering.

The DTMF generator is powered down whenever a tone is not being generated, by setting DTMFMODE[1:0] to "00".

## TRANSMIT & RECEIVE SAT PATHS

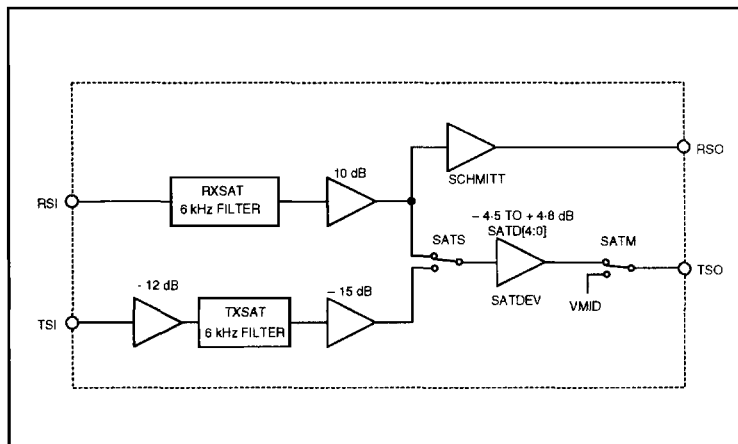


Fig. 13 Transmit and Receive SAT Paths.

### Re-transmitted SAT

ACE9040 provides two alternative paths for Supervisory Audio Tones (SAT). The first of these re-transmits the received SAT tone to the base station after narrow band filtering and providing signal level adjustment. This path is selected by setting SATS bit to "0" in the "Initializing mode 1" control message. The baseband signal from the receiver FM discriminator drives the ACE9040 through the RSI pin into the RXSAT 6 kHz bandpass filter required for AMPS or TACS. The recovered SAT signal then passes through a 10 dB amplifier and the SATS selector switch to the deviation setting

amplifier SATDEV. This is controlled by bits SATD[4:0] in the "Initializing mode 0" control message. SAT then passes through the SATM mute switch, controlled by bit SATM in the "Operation mode" control message, to output pin TSO for input to the modulation combiner.

### Regenerated SAT

The alternative SAT path externally measures the SAT frequency and generates a local tone to match. This route is

## ACE9040

selected by setting the SATS bit to "1" in the "Initializing mode 1" control message. The ACE9040 connects the receive filter RXSAT output through a Schmitt trigger to drive pin RSO with a logic level version of the received SAT. A system controller, such as an ACE9050, detects the frequency and generates a digital signal to drive back into the ACE9040 on pin TSI. After the signal level is reduced by -12 dB a 6 kHz bandpass filter TXSAT converts this square-wave into a sinewave. This is followed by a -15 dB attenuator to reduce the near logic level signal to a normal modulation level. This signal drives the same SATDEV deviation setting stage and mute switch

SATM as the returned signal to give an output at TSO.

### Base Station Originated SAT

ACE9050, System Controller, can be used to generate a squarewave SAT at 6 kHz which is input to pin TSI and filtered by the TXSAT filter and output at TSO. The RXSAT filter path with its output at RSO, can be used to filter the received SAT from a mobile for verification by an external frequency detector that the mobile is transponding the correct tone. This is the same as the regeneration loop above but starting with generation.

## TRANSMIT SIGNAL COMBINER

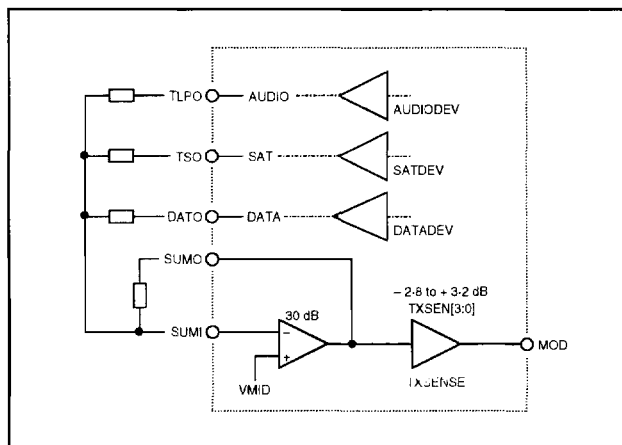


Fig. 14 Transmit Signal Combining Network and Modulation Driver

Used to modulate the transmitted r.f. output, the speech and optional DTMF signals at TLPO, SAT at TSO, and data and ST at DATO, are combined using an internal op-amp. This op-amp has an inverting input at pin SUMI and output at pin SUMO, the non-inverting input is internally biased to VMID. With an external feedback resistor between SUMI and SUMO, external resistors sum the inputs into pin SUMI and are chosen for each different cellular system to select the relative and absolute gains to give the correct deviation for each component of the modulation. Individual fine adjust-

ments to take out component value tolerances can be made by setting: AUDIODEV, SATDEV and DATADEV gains, described in more detail in the sections TRANSMIT VOICE PATH, TRANSMIT AND RECEIVE SAT PATHS, and TRANSMIT DATA AND DTMF PATHS. A fine adjustment is made to the combined signal level by TXSENSE which drives the modulator through pin MOD. The gain of TXSENSE is set by bits TXSEN[3:0] in the "Initializing mode 1" control message over the range -2.8 to +3.2 dB.



## RECEIVE VOICE PATH

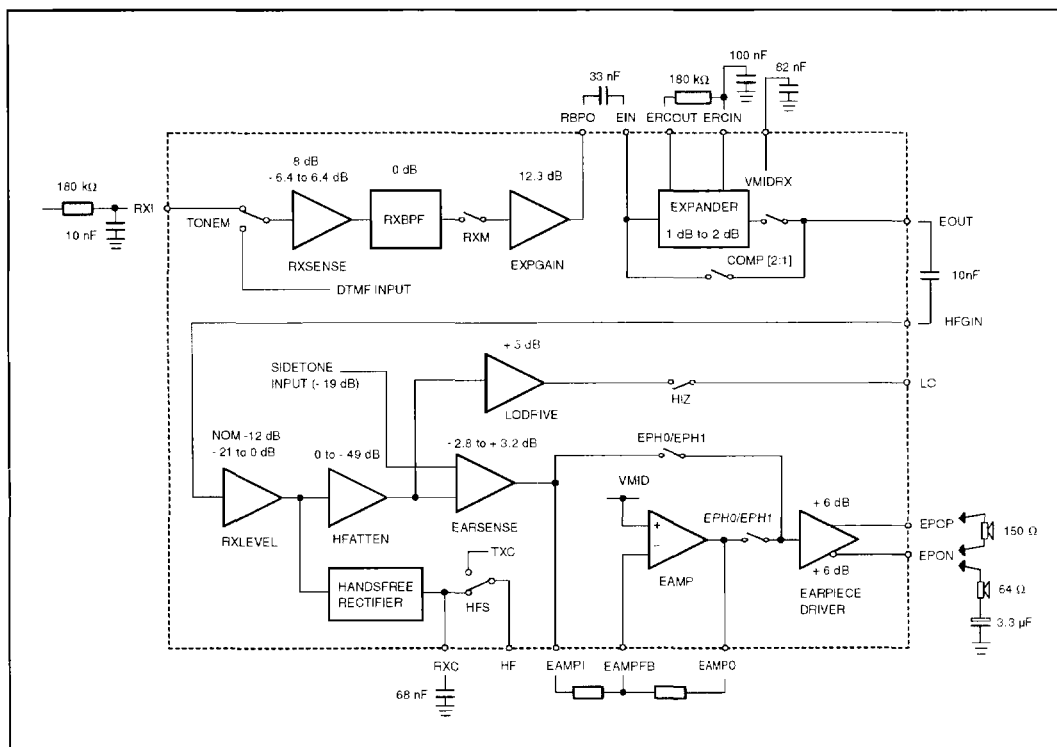


Fig. 15 Receive Voice Path

**De-emphasis and Receive Signal Input**

Demodulated FM signals drive the RXI input pin via an external de-emphasis lowpass R-C filter of typically 180 kΩ and 10 nF. With TONEM switch set to RXI the input signal is amplified in the block RXSENSE with a gain of +8 dB. RXSENSE also provides fine adjustment over a range of -6.0 dB to +6.4 dB to take up signal level tolerances in the receiver output. Fine gain adjustment is controlled by RXSEN[4:0] bits in the "Initializing mode 0" control message.

When the DTMF generator in the transmit section is in use its output can be switched into the receive path to replace the RXI signal by setting bit TONEM in the "Operation mode" control message to "1". This does not affect the transmitted signal but allows the user to hear DTMF tones to confirm key press operation.

**RX Bandpass Filter**

The RXSENSE amplifier's output is bandpass filtered to the speech bandwidth of 300 to 3400 Hz by receive bandpass filter RXBPF, as shown in figure 7. RXBPF uses switched capacitor filter techniques but does not include an anti-alias input filter as signals at RXI from the external receiver's output and the internal DTMF generator's output are already bandlimited. The F.M. discriminator output signals from ACE9040's companion device "ACE9030: Radio Interface

and Twin Synthesiser" are bandlimited by its output filter and ACE9040's internal DTMF tones are generated as sinewaves without the need for a further anti-aliasing filter.

RXBPF filter output passes through the receive mute switch controlled by bit RXM in the "Operation mode" control message and is buffered to drive pin RBPO by amplifier EXPGAIN. During mute the RBPO pin is driven to the signal ground voltage at mid supply (as found on pin BIAS). EXPGAIN gain is nominally 12.3 dB when using the internal expander and 0 dB when using an external expander. The output circuit driving RBPO includes a smoothing filter to remove clock noise.

**Expander**

Input to the expander at pin EIN is coupled by an external capacitor from RBPO to remove any d.c. voltage offsets. Using external coupling also allows the option of using an external compander or bypassing the expander if a linear system is required. In either case the signal should feed back into ACE9040 at pin HFGIN. ACE9040's compander can be bypassed by setting bits COMP[2:1] in the "Initializing mode 0" control message to "00".

A ring tone from the BAR (Beep, Alarm, Ring) generator of ACE9050 "System Controller" can be added to the ex-

## ACE9040

pander input EIN by using an external summing network and internally open circuiting the drive to pin RBPO during the tone (not just muting the speech). This is achieved with bits COMP[2:1] in the "Initializing mode 0" control message set to "01".

Signal dynamic range at input pin EIN is doubled in the 1:2 expander to restore the original signal. Within the operating signal range each 1 dB change in input level gives a 2 dB change in output level. The expander output drives pin EOUT which is coupled by an external capacitor to the input pin HFGIN. The external connection allows use of an external compander and removes any d.c. voltage offsets. Bits COMP[2:1] in the "Initializing mode 0" control message can be used to select external companding mode and power down the internal compressor and expander.

The expander's unity gain (unaffected) level EIN to EOUT is 1 V. Above this level gain is applied to the signal at EIN and below this level the signal is attenuated to achieve 1:2 dB expansion. Table 21 gives nominal, Vmax and Vmin levels at EIN and nominal levels at EOUT corresponding with the TACS and AMPS systems for 0 dB, maximum and minimum deviation.

Expander pins ERCIN and ERCOUT are used to set the attack and decay times for the expansion process. For the usual attack time of 3.0 ms and decay time of 13.5 ms, a resistor of 180 k $\Omega$  is connected between ERCIN and ERCOUT pins and a capacitor of 100 nF from ERCIN to GND. An 82 nF capacitor should be connected between VMIDRX and GND. Attack and decay time is measured with a 6 dB step, -4 dB to -10 dB relative to the unaffected level. Attack and decay times are defined respectively at points on the output envelope where it reaches x 0.57 and x 1.5 of the final steady state level.

### Volume Control and Handsfree Attenuator

Two variable gain stages follow the expander, RXLEVEL for the volume control and HFATTEN for use with handsfree mode. Both blocks provide attenuation, expressed as gain to assist system level design, RXLEVEL from 0 to -21 dB and HFATTEN from 0 to -49 dB. RXLEVEL is controlled by bits RXV[2:0] in the "Operation mode" control message and HFATTEN is controlled by bits RHF[2:0] in the "Handsfree mode" control message.

### RX Audio Output: Line Output and Earpiece

Following the handsfree attenuator the signal path splits into two parallel paths: a line output for loudspeaking phones and drivers for a dynamic earpiece or external handsfree earpiece. Bits EPH1 and EPH0 in the "Operation Mode" and "Initializing Mode 0" control messages respectively control the operation of these outputs, see table 5.

The earpiece output path begins with a variable gain stage EARSENSE which is controlled by bits EARS[3:0] in the "Initializing mode 1" control message to give a gain range of -2.8 to +3.2 dB. A sidetone signal from the output of the handsfree attenuator HFGAIN at pin TBPO is added at EARSENSE's input if bit SD in the "Initializing mode 0" control

message is set to "1". The output of this block is at pin EAMPI.

The signal at pin EAMPI is amplified by an opamp whose gain is set by external resistors, allowing overall gain setting for different models of cellular terminal. A resistor is connected from EAMPI to the amplifier input pin EAMPFB and a feedback resistor is connected from the amplifier output EAMPO to EAMPFB. The ratio of these two resistors sets the gain and the opamp's output including feedback resistors should not be loaded with less than 15 k $\Omega$ . Additional filtering can also be added to the receive path using the EAMP opamp.

### Earpiece

The earpiece drivers have outputs at pins EPOP and EPON. One of three modes of output drive or a muted output condition is selected by bits EPH1 and EPH0, see table 5.

A dynamic earpiece, typically of 150  $\Omega$  resistance in series with 800  $\mu$ H, can be driven when connected between pin EPON and EPOP. The drive mode can be either differential (EPH1 = 1, EPH0 = 1) or single ended (EPH1 = 1, EPH0 = 0). The differential output, drives a minimum of 4 Vpp into the load. The single ended output appears at EPON and drives a minimum of 2 Vpp into a load referenced to pin EPOP, which provides an output voltage at mid-supply.

Output drive is setup for an external handsfree earpiece with EPH1 = 0 and EPH0 = 1. This load, typically of 64  $\Omega$  resistance in series with 3.3  $\mu$ F, is driven from pin EPON to ground and the EPOP output is put into a high impedance state. The minimum drive into this load is 1.1 Vpp.

The input for the EPOP and EPON output drivers is driven by the signal at the EAMPO pin for single ended and differential outputs or from EAMPI directly for a handsfree earpiece, bypassing the EAMP opamp. The gain from the earpiece drivers' common input, to both EPON's inverting and and EPOP's non-inverting outputs, is nominally +6 dB.

### Line Output

Line output amplifier LODRIVE with its output at pin LO has a gain of +5 dB and is used to drive an external audio power amplifier. LODRIVE can drive a 1 k $\Omega$  load with a minimum of 1 Vrms. The LO output can be put into a high impedance state by setting bit HIZ in the "Operation mode" control message to "0". During power down EPON and EPOP and LO are tied to mid-supply voltage.

### HANDSFREE OPERATION

In a handsfree telephone the simplest method of preventing howl round caused by acoustic feedback is to attenuate either the forward or return path until the loop gain is too low for sustained oscillation. The least active path is attenuated so the signal level in each path must be detected and compared so that the quieter can be attenuated.

In the ACE9040 the signal level in both the transmit and the receive paths are rectified, with smoothing capacitors at pins TXC and RXC respectively, to give d.c. voltages corresponding to the signal levels.

A switch HFS with its output at pin HF can be internally toggled between TXC and RXC to allow measurement of the

System	Vmax at EIN		Vmin at EIN		Nominal Input at EIN (0dB) levels mVrms	Nominal Output at EOUT levels mVrms
	Vrms	dB	mVrms	dB		
TACS	1.0	+ 12.5	43	- 15	245	60
AMPS	1.0	+ 12.3	22	-21	245	60

Table 21 Expander EIN and EOUT signal levels for TACS and AMPS

two levels at these pins by an external level sensing circuit such as an analog to digital converter input of ACE9030. The HFS switch is controlled by bit HFS in the "Handsfree mode" control message.

The system controller after comparison of the voltage levels at TXC and RXC pins can attenuate the weaker signal path by up to 49 dB, in 7 dB steps using blocks HFGAIN for transmit and HFATTEN for receive. Bits THF[2:0] and RHF[2:0] in the "Handsfree mode" control message are used to set the gains of HFGAIN and HFATTEN respectively. The rate of change of gain should be limited in the system controller to allow normal conversation.

Attack and decay time constants are set by the resistance and capacitance on the TXC and RXC pins. With the internal resistor to ground of approximately 500 k $\Omega$  and an external capacitor to ground of 68 nF the normal attack time of 1 ms and decay time of 35 ms is achieved. By adding a parallel resistor the ratio of attack to decay time can be altered.

To save power in a hand portable when handsfree operation is not needed, the transmit and receive signal rectifiers can be switched off by setting bit HFP in the "Handsfree mode" control message to "0".

## BIASES AND REFERENCES

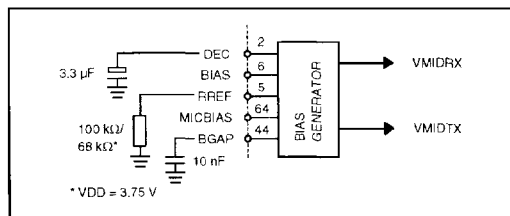


Fig. 16 Bias Circuits

### BIAS, VMID and MICBIAS

Within ACE9040 most signals are single ended and swing either side of a mid-supply reference voltage. These internal references are all labelled VMIDxx in this data sheet.

A low impedance voltage source at mid-supply for use as an external signal ground is available on pin BIAS. This is a buffered copy of the voltage at pin DEC which is from an internal high impedance potential divider between  $V_{DD}$  and

$V_{SS}$ . The DEC pin should be decoupled to ground with a capacitor of greater than 3.3  $\mu$ F. Two additional buffers provide copies of DEC's voltage at pins VMIDTX and VMIDRX, these are used as internal signal grounds for the transmit and receive paths respectively. VMIDTX and VMIDRX pins should be decoupled to GND with 82 nF capacitors. By using separate mid-supply signal grounds crosstalk due to the compander time constant circuits and the speech and tone signals are kept to a minimum.

Pin MICBIAS gives the bias needed for an electret microphone nominally 0.8 times  $V_{DD}$ , e.g. when  $V_{DD}$  is 3.75 V MICBIAS = 3 V

### OP-AMP Reference Current

Reference currents for all the internal op-amps are set by an external resistor connected from pin RREF to ground ( $V_{SS}$ ). Nominal values are 100 k $\Omega$  for  $V_{DD}$  = 4.85 V and 68 k $\Omega$  for  $V_{DD}$  = 3.75 V. A stable discrete resistor should be used to ensure consistent operation over a wide temperature range.

### Power Supply Comparator - Reset Output

A power supply comparator is provided to give a reset at power-on and enable the system controller to initiate a clean shut-down sequence if the battery voltage falls too low. When  $V_{DD}$  is below a band-gap derived threshold the open-drain output pin LVN drives to a logic low. This occurs for  $V_{DD}$  exceeding 1 V but less than a typical threshold of 3.35 V. An external resistor at LVN provides a pull-up to  $V_{DD}$  with a capacitor to ground ( $V_{SS}$ ) to give a power-on reset delay. Typical values for RC are 220 k $\Omega$  and 150 nF. This RC combination also removes short transients or noise pulses from the signal at LVN during power up. If this comparator is not required the bandgap and comparator can be powered down by setting bit PDLVC in the "Initializing mode 0" control message to a "1".

### Serial Data Clock

All switched capacitor filter switching clocks are derived from the serial data clock SCLK which must be fixed at 1.008 MHz to ensure correct frequency responses.

### AMPLIFIER

An uncommitted op-amp is provided with its non-inverting input internally connected to VMID, inverting input at pin AMPI and output at pin AMPO.

## ACE9040

### APPLICATIONS INFORMATION

To help with system set up tables 22 to 25 show ACE9040's functions and their respective controlling bits. Table 23 shows the gains and filter characteristics pre-determined when setting the TACS & AMPS cellular system

selection bits. Tables 23 & 24 show these functions respectively for the transmit and receive sections of ACE9040. Table 25 shows the four control messages with an example of the data to turn all ACE9040 functions on.

Function Controlled by TACS & AMPS Bits			
		TACS	AMPS
Internal pre-emphasis gain control.	TACS = - 7.0 dB AMPS = - 11.0 dB	1 0	0 1

Table 22 Functions Controlled by TACS & AMPS Bits

Gain and filter set-ups for TACS, AMPS and user control bits: Transmit			
	TACS	AMPS	User control bit(s)
Microphone amp	22 or 32 dB	22 or 32 dB	MIG
Soft limiter	0 to - 30 dB	0 to - 30 dB	Internal
INPSENSE gain adjust	+ 8 dB (- 12.8 to + 12 dB)	+ 8 dB (- 12.8 to + 12 dB)	INPS[4:0]
TXBPF TX bandpass filter	0 dB	0 dB	Fixed
HFGAIN, TX Handsfree attenuator	0 to - 49 dB	0 to - 49 dB	THF[2:0]
Compressor: Compression about unaffected level. (707 mVrms, - 3 dBV)	2:1 <sup>1</sup>	2:1 <sup>1</sup>	Fixed
Pre-emphasis gain control (Internal compressor)	- 7 dB	- 11 dB	TACS & AMPS
Pre-emphasis gain control	0 dB	0 dB	Fixed
HARD LIMIT, Hard deviation limiter	+ 16.5 dB @ 4.85V + 13.5 dB @ 3.75V	+ 13.5 dB @ 3.75V + 16.5 dB @ 4.85V	V485 pin
TXLPF TX low pass filter	- 8 dB	- 8 dB	Fixed
AUDIODEV,	0 dB (-2.8 to + 3.2 dB)	0 dB (-2.8 to + 3.2 dB)	AUDEV[3:0]
TXSENSE, signal	0 dB (-2.8 to +3.2 dB)	0 dB (-2.8 to +3.2 dB)	TXSEN[3:0]
Nominal TX Channel Gain (bypassed compressor)	28.5 <sup>2</sup> dB @ 3.75 V 31.5 <sup>2</sup> dB @ 4.85 V	23.5 <sup>2</sup> dB @ 3.75 V 26.5 <sup>2</sup> dB @ 4.85 V	

Notes: 1. Above the unaffected (0 dB gain) level the compressor attenuates and below this level it provides gain.  
2. MIC gain = 22 dB

Table 23 Transmit gain and filter set-ups for TACS, AMPS and user control bits

Gain and filter set-ups for user control bits: Receive			
	TACS	AMPS	User control bit(s)
External De-emphasis	- 21 dB <sup>1</sup>	- 21 dB <sup>1</sup>	Fixed externally
RXSENSE: Receive audio gain	+ 8 dB	+ 8 dB	Fixed
RXSENSE: Receive audio gain adjustment range	- 6 to + 6.4 dB	- 6 to + 6.4 dB	RXSEN[4:0]
RXBPF: RX bandpass filter	0 dB	0 dB	Fixed
RX Expander gain EXPGAIN: Internal	+ 12.3 dB	+ 12.3 dB	Fixed
RX Expander gain EXPGAIN: External	0 dB	0 dB	Fixed
Expander: Expansion about unaffected level (1000 mV, 0 dBV)	1:2 <sup>2</sup>	1:2 <sup>2</sup>	Fixed
RXLEVEL, Receive volume control	- 12 dB (+ 12 to - 9 dB)	- 12 dB (+ 12 to - 9 dB)	RXV[2:0]
HFATTEN, RX handsfree attenuator	0 to - 49 dB	0 to - 49 dB	RHF[2:0]
EARSENSE	0 dB (- 2.8 to + 3.2 dB)	0 dB (- 2.8 to + 3.2 dB)	EARS[3:0]
Ear piece driver	6 dB	6 dB	Fixed
Nominal Receive Gain	- 6.7 dB	- 6.7 dB	Fixed
(Expander bypassed)			
DATA and SAT Filter:s			
TX data path filter cut-off	16 kHz	20 kHz	DATAF[1:0]
TX & RX bandpass SAT filter centre frequency	6 kHz	6 kHz	Fixed

Notes: 1. Attenuation with an external de-emphasis network of series 190 k $\Omega$  with 10 nF to GND at RX1 input.  
2. Above the unaffected (0 dB gain) level the expander provides gain and below this level it attenuates.

Table 24 Receive, Data and SAT gain and filter set-ups for user control bits

OPERATING MODE								
WORD/BIT	D7	D6	D5	D4	D3	D2	D1	D0
DATA 1	PREEMPH	MIS	DTMF3	DTMF2	DTMF1	DTMF0	TONEM	DTMFM
DATA 2	DTMFMODE1	DTMFMODE0	DTMFEN	MLI	TXM	DATM	SATM	RXM
DATA 3	EPH1	HIZ	RXV2	RXV1	RXV0	PD	0	1
START-UP BIT SETTINGS								
DATA 1	0	0	0	0	0	0	0	0
00 <sub>HEX</sub>								
DATA 2	0	0	0	0	1	1	1	1
0F <sub>HEX</sub>								
DATA 3	1	1	1	1	1	0	0	1
F9 <sub>HEX</sub>								

INITIALIZING MODE 0								
WORD/BIT	D7	D6	D5	D4	D3	D2	D1	D0
DATA 1	SOFTLIMIT	SD	SATD4	SATD3	SATD2	SATD1	SATD0	RXSEN4
DATA 2	RXSEN3	RXSEN2	RXSEN1	RXSEN0	INPS4	INPS3	INPS2	INPS1
DATA 3	INPS0	COMP2	COMP1	PDLVC	DTWIST	EPH0	0	0
START-UP BIT SETTINGS								
DATA 1	1	0	1	0	0	0	0	1
A1 <sub>HEX</sub>								
DATA 2	0	0	0	0	1	0	0	0
08 <sub>HEX</sub>								
DATA 3	0	1	0	0	1	1	0	0
4C <sub>HEX</sub>								

INITIALIZING MODE 1								
WORD/BIT	D7	D6	D5	D4	D3	D2	D1	D0
DATA 1	EARS3	EARS2	EARS1	EARS0	AUDEV3	AUDEV2	AUDEV1	AUDEV0
DATA 2	DATD3	DATD2	DATD1	DATD0	TXSEN3	TXSEN2	TXSEN1	TXSEN0
DATA 3	TACS	DATAF1	DATAF0	SATS	TEST1	TEST0	1	0
START-UP BIT SETTINGS (TACS)								
DATA 1	1	0	0	0	1	0	0	0
88 <sub>HEX</sub>								
DATA 2	1	0	0	0	1	0	0	0
88 <sub>HEX</sub>								
DATA 3	1	1	0	0	0	0	1	0
C2 <sub>HEX</sub>								

HANDSFREE								
WORD/BIT	D7	D6	D5	D4	D3	D2	D1	D0
DATA 1	X	X	X	X	X	0	0	0
DATA 2	0	X	AMPS	X	X	MIG	HFS	HFP
DATA 3	RHF2	RHF1	RHF0	THF2	THF1	THF0	1	1
START-UP BIT SETTINGS								
DATA 1	X	X	X	X	X	0	0	0
00 <sub>HEX</sub>								
DATA 2	0	X	0	X	X	0	0	1
01 <sub>HEX</sub>								
DATA 3	0	0	0	0	0	0	1	1
03 <sub>HEX</sub>								

Table 25 Control Messages