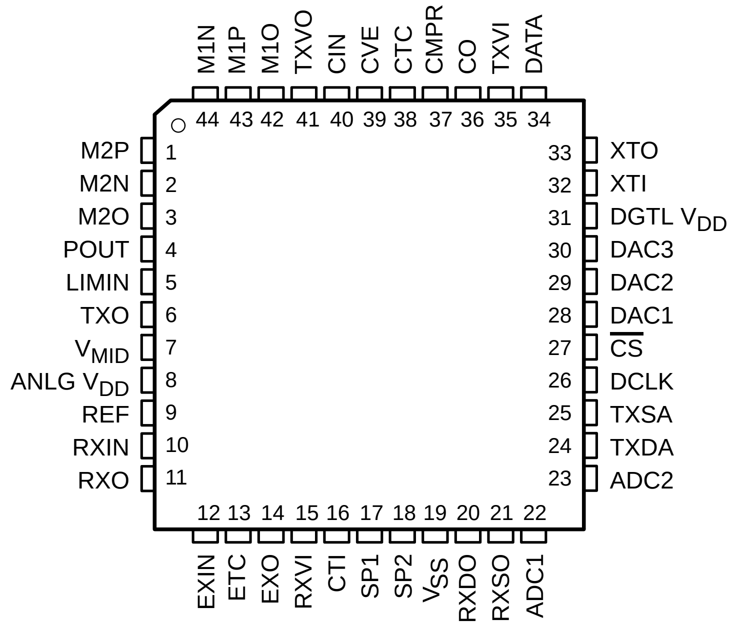
TCM8010-50
AMPS/TACS AUDIO PROCESSOR

SLWS009A – SEPTEMBER 1994 – REVISED DECEMBER 1995

- **TACS and AMPS Operation**
- **Integrated RX and TX Voice Filters**
- **Integrated RX and TX Data Filters**
- **RX and TX Narrow-Band SAT Filters**
- **RX Data Recovery Comparator**
- **Preemphasis and Deemphasis Filtering**
- **Adjustable TX Limiter**
- **Microphone Preamplifiers**
- **Digitally Controlled Gains and Signal Selection or Muting**
- **Three 8-Bit DACs With Output Buffers**
- **8-Bit ADC With Input Multiplexer**
- **DTMF Generator**
- **On-Chip Compandor**
- **Flexible Clock or Oscillator Operation**
- **Simple 3-Wire Digital Interface**
- **Low-Power, 2-mA Standby Current**
- **44-Pin Mini QFP (FR) Package**

FR PACKAGE
(TOP VIEW)



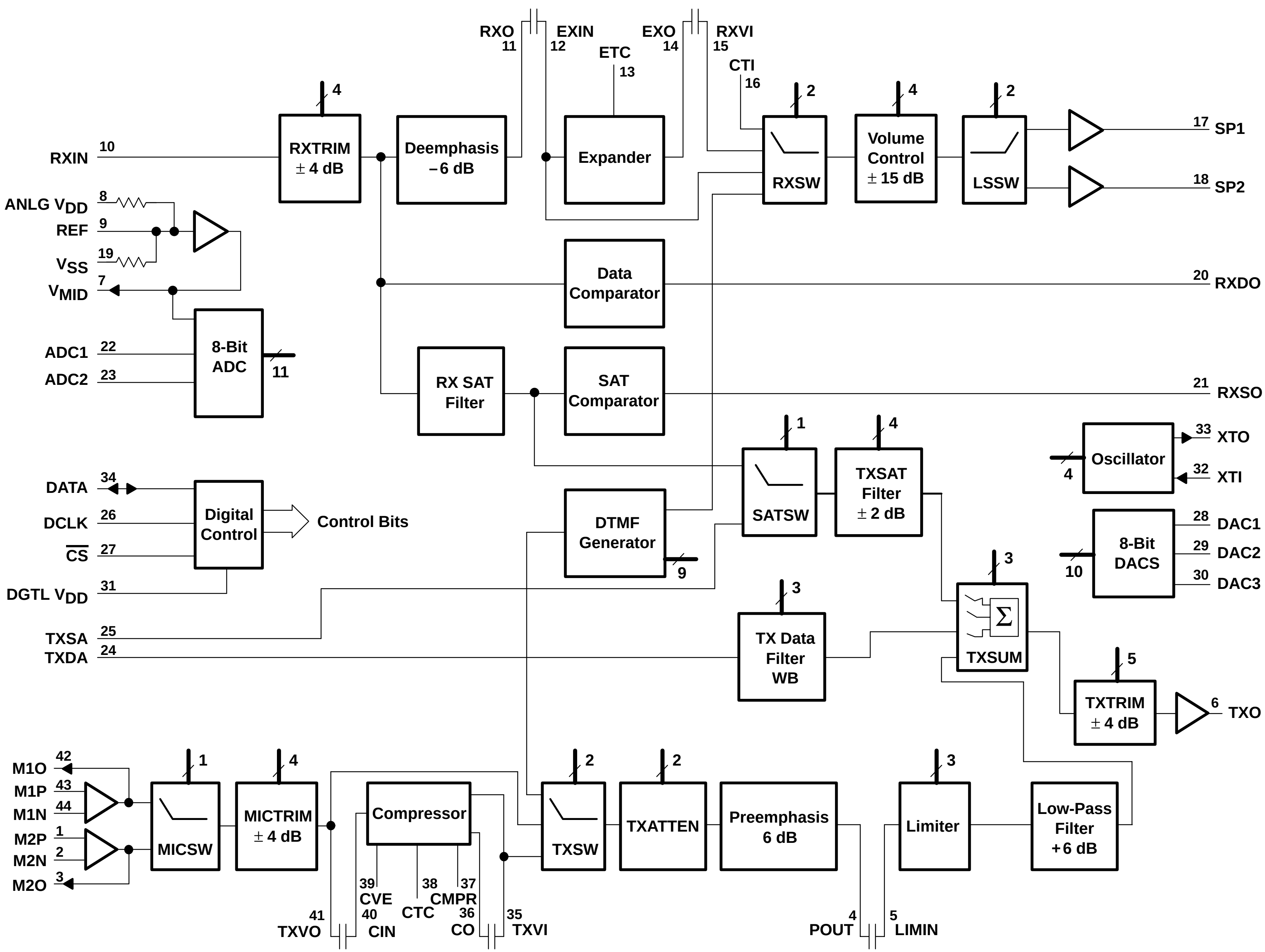
description

The TCM8010-50 is a complete advanced mobile phone service (AMPS) / total access communications system (TACS) audio processor built using the Texas Instruments Advanced LinBiCMOS™ technology and packaged in a 44-pin mini QFP (FR) package. This device provides a highly integrated solution for analog-signal processing in mobile and hand-held FM cellular telephones while conserving circuit board area and vertical height within the finished product. All necessary voice and data filters, and all appropriate antialiasing and smoothing filters are incorporated in the device. Continuous-time filters are used for the anti-aliasing and smoothing functions and switched-capacitor techniques are used only where appropriate. Ancillary functions such as microphone preamplifiers, differential loudspeaker outputs, CCITT-compatible compander, dual tone multi-frequency (DTMF) generator, three 8-bit digital-to-analog converters (DACs), and an 8-bit analog-to-digital converter (ADC) with input multiplexer are also included in the device. A simple 3-wire serial interface provides digital control of signal-path switching, muting and gain adjustment, the 8-bit DACs, transmit (TX) limit level, DTMF code and amplitude, ADC multiplexer input select, and allows the ADC output to be read.

In active mode, the TCM8010-50 consumes less than 12 mA of supply current. When the DTMF generator or the ADC are not in operation, the power consumption is even less. The device can be put into a standby mode in which only the receive (RX) data path is active, reducing the supply current to a typical value of 2 mA or less.

Either the integrated-clock oscillator or an external clock signal (with several frequency options) can be used.

functional block diagram



Terminal Functions

TERMINAL NAME	TERMINAL NO.	I/O	DESCRIPTION
ADC1 – ADC2	22, 23	I	ADC input 1 and 2 (analog)
ANLG V_{DD}	8		Analog positive supply
CIN	40	I	Compressor input (analog)
CMPR	37	I	Compressor rectifier input (analog)
CO	36	O	Compressor output, ac coupled to CMPR and to TXVI (analog)
$\overline{CS}$	27	I	Serial interface chip select, active low (digital)
CTC	38	O	Compressor time constant (analog)
CTI	16	I	Call tone input (analog and digital)
CVE	39	I	Compressor virtual ground (analog)
DAC1 – DAC3	28, 29, 30	O	DAC outputs (analog)
DATA	34	I/O	Serial interface data signal (digital)
DCLK	26	I	Serial interface clock signal (digital)
DGTL V_{DD}	31		Digital positive supply
ETC	13	O	Expander time constant (analog)
EXIN	12	I	Expander input (analog)
EXO	14	O	Expander output, ac coupled to RXVI (analog)
LIMIN	5	I	Limiter input (analog)
M10	42	O	Microphone preamplifer 1 output (analog)
M1P/N	43, 44	I	Microphone preamplifier 1 differential inputs (analog)
M2O	3	O	Microphone preamplifer 2 output (analog)
M2P/N	1, 2	I	Microphone preamplifier 2 differential inputs (analog)
POUT	4	O	Preemphasis output, ac coupled to LIMIN (analog)
REF	9		Midrail reference – decouple to V_{SS} with external capacitor
RXDO	20	O	Receive section data output (digital)
RXIN	10	I	Receive section input (analog)
RXO	11	O	Receive section deemphasis voice filter output (analog)
RXSO	21	O	Receive section supervisory audio tone (SAT) output (digital or analog)
RXVI	15	I	Voice input to volume control stage (analog)
SP1/2	17, 18	O	Speaker outputs 1 and 2 (analog)
TXDA	24	I	Transmit data filter input (digital or analog)
TXO	6	O	Transmit section output (analog)
TXSA	25	I	Transmit SAT input (digital or analog)
TXVI	35	I	Input to TX voice-path output stages (analog)
TXVO	41	O	Transmit voice input stage output, ac coupled to CIN (analog)
V_{MID}	7		Buffered midrail voltage – decouple to V_{SS} with external capacitor
V_{SS}	19		Negative supply (0 V)
XTI/XTO	32, 33		Crystal oscillator and clock recovery inputs

absolute maximum ratings over operating free-air temperature range†

Supply voltage range, V_{DD} (see Note 1) . . . . –0.3 V to 7 V
Input voltage, V_I (any pin) . . . . V_{SS} – 0.3 V to V_{DD} + 0.3 V
Operating free-air temperature range, T_A . . . . 0°C to 70°C
Continuous total power dissipation at (or below) T_A = 25°C . . . . 893 mW
Storage temperature range, T_{stg} . . . . –65°C to 150°C

† Stresses beyond those listed under “absolute maximum ratings” may cause permanent damage to the device. These are stress ratings only, and functional operation of the device at these or any other conditions beyond those indicated under “recommended operating conditions” is not implied. Exposure to absolute-maximum-rated conditions for extended periods may affect device reliability.

NOTE 1: All voltages are with respect to V_{SS}.

recommended operating conditions

	MIN	NOM	MAX	UNIT
Supply voltage, DGTL V_{DD} and ANLG V_{DD}	4.5	5	5.5	V
High-level input voltage, V_{IH}	$0.8V_{DD}$			V
Low-level input voltage, V_{IL}			0.8	V
Operating virtual junction temperature, T_J	–30		70	°C

electrical characteristics over recommended operating virtual junction temperature range, V_{DD} = 5 V, f_{xtal} = 2.56 MHz

PARAMETER			MIN	TYP	MAX	UNIT
$I_{DD(A)}$	Analog supply current	Standby mode, DACs off		1	1.7	mA
		Standby mode, DACs on		1.4	2	
		Operating mode		11	16	
		Including DTMF generator		12	17	
$I_{DD(D)}$	Digital supply current	Standby mode		160	1000	µA
		Operating mode		0.5	1.7	mA
		ADC operating		1		
REF	Midsupply reference voltage	Operating mode	2.4	2.5	2.6	V
V_{MID}	Buffered midsupply reference voltage	Operating mode	2.4	2.5	2.6	

analog inputs

PARAMETER		MIN	TYP	MAX	UNIT
I_I	Input current at M1P, M1N, M2P, M2N, ADC1, ADC2, CTI		1		µA
z_i	Input impedance at RXIN, RXVI, LIMIN, TXSA, TXDA	100			kΩ
	Input impedance at EXIN, CIN, CMPR, TXVI	25			

digital interface

PARAMETER		TEST CONDITIONS	MIN	TYP	MAX	UNIT
I_{IH}	High-level input current	V_I = 5 V			1	µA
I_{IL}	Low-level input current	V_I = 0 V			1	
f_{CLK}	Serial clock frequency, DCLK input				1	MHz
V_{OH}	High-level output voltage	I_{OH} = 500 µA	$0.9V_{DD}$			V
V_{OL}	Low-level output voltage	I_{OL} = 500 µA			$0.1V_{DD}$	

transmit path electrical characteristics

input stage gain M1O/M2O to TXVO, V_{DD} = 5 V

PARAMETER	TEST CONDITIONS	MIN	MAX	UNIT
Gain	MICTRIM = <1000> (see Note 2)	–0.5	0.5	dB
MICTRIM positive range	MICTRIM = <1111>	3.3	4.3	dB
MICTRIM negative range	MICTRIM = <0000>	–4.8	–3.8	dB
MICTRIM step size		0.38	0.68	dB
Preamp CMRR		48		dB
Distortion	V_I = 1 V, f = 1 kHz		0.5%	
MICSW isolation	V_I = 100 mV, f = 1 kHz	50		dB

NOTE 2: The control bits associated with a block or function are shown in < >.

compressor CIN to CO, V_{DD} = 5 V

PARAMETER	TEST CONDITIONS	MIN	TYP	MAX	UNIT
Unity gain level†		76	103	127	mV
Relative linearity error	$V_I = V_{ref}$ + 2 dB to V_{ref} –18 dB		–0.01	±0.5	dB
	$V_I = V_{ref}$ – 18 dB to V_{ref} –48 dB		–0.16	±1	dB
R_{COMP} compressor resistance		37	47	67	kΩ

† This parameter becomes V_{ref} for the relative-linearity-error test conditions.

output stage TXVI to TXO, V_{DD} = 5 V

PARAMETER	TEST CONDITIONS	MIN	MAX	UNIT
TXTRIM step size		0.16	0.36	dB
TXTRIM positive range	TXTRIM = <11111>	3.5	4.5	dB
TXTRIM negative range	TXTRIM = <00000>	–4.8	–3.8	dB
TXATTEN step size		7	9	dB
TXATTEN range		21	27	dB

output stage limiter TXVI to TXO, V_{DD} = 5 V

PARAMETER	TEST CONDITIONS	MIN	MAX	UNIT
Maximum output signal	TXVI = 316 mV, f = 300 Hz to 25 000 Hz, LIM = <110>		1900	mVp-p
Distortion	f = 1 kHz, level at TXO = 2/3 × level measured in previous test, LIM = <110>		3%	
Trim step size, analog test mode A, output at RXDO	TXVI = 316 mV	0.8	1.2	dB
Trim positive range, analog test mode A, output at RXDO	LIM = <111>	2.5	3.5	dB
Trim negative range, analog test mode A, output at RXDO	LIM = <000>	–4.5	–3.5	dB

output stage frequency response TXVI to TXO, V_{DD} = 5 V

PARAMETER	TEST CONDITIONS		MIN	MAX	UNIT
Frequency response	0-dB reference at f = 1 kHz, TXVI = 26 mV	f < 200 Hz		–20	dB
		f = 300 Hz	–13.46	–9.46	dB
		f = 500 Hz	–9.02	–5.02	dB
		f = 2000 Hz	3.02	7.02	dB
		f = 2500 Hz	4.96	8.96	dB
		f = 3000 Hz	4.96	10.54	dB
		f = 5900 Hz		–35	dB
		f = 6000 Hz		–35	dB

overall transmit path electrical characteristics M1O/M2O to TXO, TXATT = <00>, V_{DD} = 5 V

PARAMETER	TEST CONDITIONS	MIN	TYP	MAX	UNIT
Compressor bypass gain	MICT = <1000>, TXT = <10000>	10.8	12	13.0	dB
Output noise, compressor enabled, M1O/M2O = V_{MID} psophometric weighting	RXIN = 400 mV, f = 1 kHz		2.3		mVrms
Voice mute attenuation	M1O/M2O = 100 mV, f = 1 kHz	50	–80		dB

DATA output levels TXDA to TXO, V_{DD} = 5 V

PARAMETER		TEST CONDITIONS	MIN	MAX	UNIT
Output level	AMPS	f_I = 10-kHz square wave, amplitude 0 V to 5 V	1070	1188	mVp-p
	TACS	f_I = 8-kHz square wave, amplitude 0 V to 5 V	1070	1188	mVp-p
Frequency response	AMPS	–3 dB relative to 1 kHz, Analog test mode B	17	22	kHz
	TACS		14.4	17.6	kHz
TX data mute attenuation			50		dB

SAT output levels TXSA to TXO, V_{DD} = 5 V

PARAMETER	TEST CONDITIONS		MIN	MAX	UNIT
Output level	ISAT = <0>, f_I = 6-kHz square wave, amplitude 0 V to 5 V		95	116	mV
SAT trim positive range			2	2.3	dB
SAT trim negative range			–2.7	–2.3	dB
SAT trim step size			0.2	0.4	dB
Frequency response	0-dB reference at f = 6 kHz, ISAT = <0>	f < 3 kHz		–35	dB
		f = 4.8 kHz		–25	dB
		f = 5.1 kHz		–20	dB
		f = 5.8 kHz	–5	0.5	dB
		f = 5.94 kHz	–0.5	0.5	dB
		f = 6.06 kHz	–0.5	0.5	dB
		f = 6.2 kHz	–5	0.5	dB
		f = 7.2 kHz		–20	dB
		f > 9 kHz		–35	dB
TX SAT mute attenuation			50		dB

SAT output level RXIN to TXO, RXT = <1000>, SAT = <1000>, TXT = <10000>, V_{DD} = 5 V

PARAMETER	TEST CONDITIONS	MIN	TYP	MAX	UNIT
Output level	ISAT = <1>, Input to RXIN = 6-kHz sine wave, amplitude 600 mV		400		mV

receive path electrical characteristics

input stage RXIN to RXO, V_{DD} = 5 V

PARAMETER	TEST CONDITIONS		MIN	MAX	UNIT
Gain	RXTRIM = <1000>		–6.3	–5.2	dB
RXTRIM positive range	RXTRIM = <1111>		3.2	4.2	dB
RXTRIM negative range	RXTRIM = <0000>		–4.5	–3.8	dB
RXTRIM step size			0.39	0.69	dB
Frequency response	0-dB reference at f = 1 kHz, RXIN = 400 mV	f <100 Hz		–28	dB
		f = 240 Hz		12.9	dB
		f = 300 Hz	8	11	dB
		f = 400 Hz	7.5	8.5	dB
		f = 2400 Hz	–8.2	–7.1	dB
		f = 3000 Hz	–12	–9	dB
		f > 5900 Hz		–40	dB

expander EXIN to EXO, V_{DD} = 5 V

PARAMETER	TEST CONDITIONS	MIN	TYP	MAX	UNIT
Unity gain level = V_{ref}†		80	100	130	mV
Relative linearity error	EXIN = V_{ref} +9.5 dB to V_{ref} –2.8 dB		–0.3	±1	dB
	EXIN = V_{ref} –2.8 dB to V_{ref} –23.8 dB		–0.8	±2	dB
R_{EXP} expander resistance		37.5	47	71.6	kΩ

† This parameter becomes V_{ref} for the relative-linearity-error test conditions.

output stage

PARAMETER		TEST CONDITIONS	MIN	MAX	UNIT
Volume control	Gain RXVI to SP1/SP2	VOL = <1000>	0.5	1.5	dB
	Positive range	VOL = <1111>	13	15	dB
	Negative range	VOL = <0000>	–16.5	–15.5	dB
	Step size		1.75	2.25	dB
CTI input	Gain to SP1/SP2	VOL = <1000>	0	2	dB
Expander bypass gain from RXIN to SP1/SP2		VOL = <1000>	–5.5	–4	dB
Output load at SP1/SP2			500		Ω
Output voltage at SP1/SP2		R_L = 500 Ω	2.5		Vp-p
Distortion at SP1/SP2, expander enabled		RXIN = 400 mV, f = 1 kHz, No load		2%	
Noise at SP1/SP2, expander bypassed		RXIN = V_{MID}, psophometric weighting		3	mV
Voice mute attenuation		RXIN = 400 mV, f = 1 kHz	50		dB

RX DATA comparator RXIN to RXDO, V_{DD} = 5 V

PARAMETER	TEST CONDITIONS		MIN	MAX	UNIT
Must-detect level	f = 4 kHz, 5 kHz, 8 kHz, and 10 kHz		210		mVp-p
Must-not-detect level				40	
Output duty cycle		RXIN = 900 mV peak to peak	47.5%	52.5%	

RX SAT frequency response RXIN to RXSO, SATDIG = <1>, V_{DD} = 5 V

PARAMETER	TEST CONDITIONS		MIN	MAX	UNIT
Frequency response	0-dB reference at f = 6 kHz	f < 3 kHz		–35	dB
		f = 4.8 kHz		–25	dB
		f = 5.1 kHz		–19	dB
		f = 5.8 kHz	–5	0.5	dB
		f = 5.94 kHz	–0.5	0.5	dB
		f = 6.06 kHz	–0.5	0.5	dB
		f = 6.2 kHz	–5	0.5	dB
		f = 7.2 kHz		–20	dB
		f > 9 kHz		–35	dB

RX SAT comparator RXIN to RXSO, SATDIG = <0>, V_{DD} = 5 V

PARAMETER	TEST CONDITIONS	MIN	TYP	MAX	UNIT
Must-detect level	f = 6 kHz	64	30		mVrms

miscellaneous block electrical characteristics

digital-to-analog converters DAC1, DAC2, and DAC3

PARAMETER	TEST CONDITIONS	MIN	TYP	MAX	UNIT
Output voltage at code 255	DACX2 = <1>	V_{DD} – 130			mV
Output voltage at code 255	DACX2 = <0>	V_{DD}/2 –100		V_{DD}/2 + 100	mV
Zero code offset			13	55	mV
Differential nonlinearity (codes 5 – 250)			0.3	1	LSB
Integral nonlinearity (codes 5 – 250)			0.3	1	LSB

analog-to-digital converter, DCLK = 160 kHz, V_{DD} = 5 V

PARAMETER	TEST CONDITIONS	MIN	TYP	MAX	UNIT
Full scale for inputs ADC1, ADC2, and V_{MID}		2.3		2.6	V
Differential nonlinearity			0.5	1	LSB
Integral nonlinearity			0.5	1	LSB
Clock rate (DCLK)				200	kHz

DTMF generator transmit levels at TXO, TXTRIM = <10000>, V_{DD} = 5 V

PARAMETER	DTTR	TEST CONDITIONS	MIN	TYP	MAX	UNIT
697-Hz tone, low tone	<0100>	AMPS mode	108	153	164.9	mV
1477-Hz tone, high tone			300	340	348.6	mV
697-Hz tone, low tone		TACS mode	61	78	88	mV
1477-Hz tone, high tone			140	175	190	mV
DTMF trim steps	<0000> – <0001>			0.4		dB
	<0001> – <0010>			0.4		dB
	<0010> – <0011>			0.4		dB
	<0011> – <0100>			0.5		dB
	<0100> – <0101>			0.5		dB
	<0101> – <0110>			0.6		dB
	<0110> – <0111>			0.6		dB
	<0111> – <1000>			0.7		dB
	<1000> – <1001>			0.7		dB
	<1001> – <1010>			0.8		dB
	<1010> – <1011>			0.9		dB
	<1011> – <1100>			1.0		dB
	<1100> – <1101>			1.1		dB
	<1101> – <1110>			1.3		dB
	<1110> – <1111>			1.5		dB
Positive range	<0100> – <1111>		7.1	9.8	12.1	dB
Negative range	<0100> – <0000>		–2.7	–1.9	–1	dB
Skew, change in level of high tone	<0100>		1.3	1.85	2.2	dB
Distortion products	<0100>	Relative to low tone		–30		dB

DTMF generator receive levels at SP1 and SP2, DTTR = <0100>, VOL = <1000>, V_{DD} = 5 V

PARAMETER	TEST CONDITIONS	MIN	MAX	UNIT
All tones	AMPS mode	58	67	mV
	TACS mode	29	35	mV
Distortion products	Relative to low tone		–40	dB

TYPICAL CHARACTERISTICS

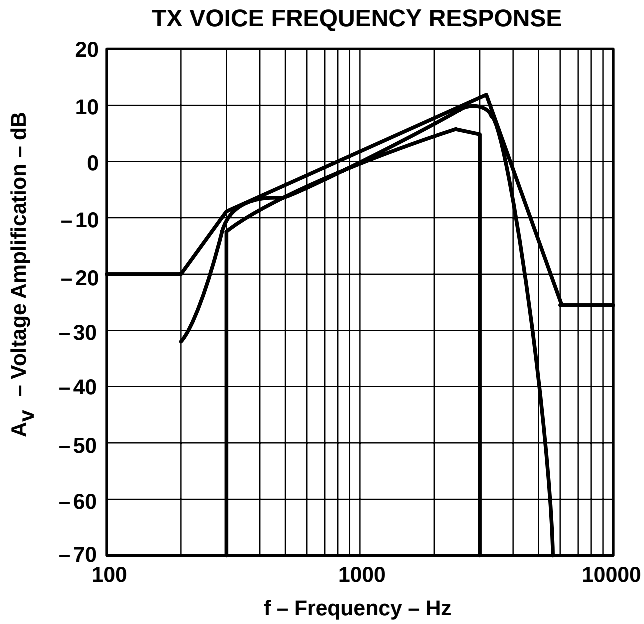


Figure 1

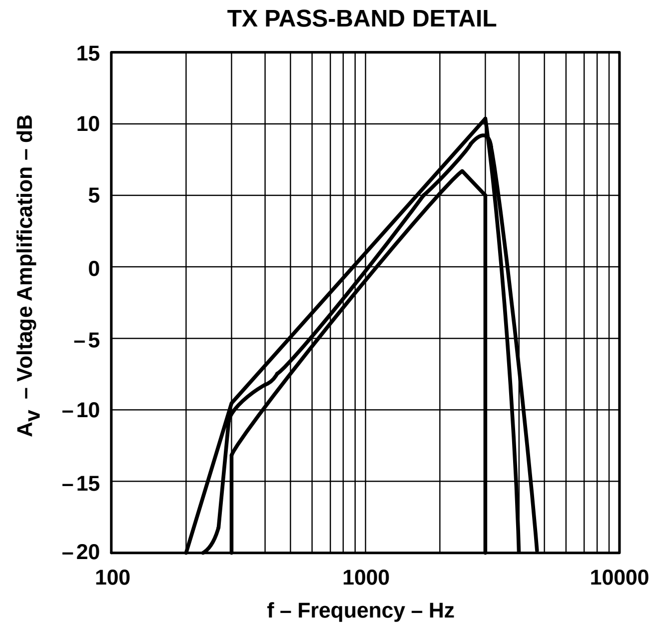


Figure 2

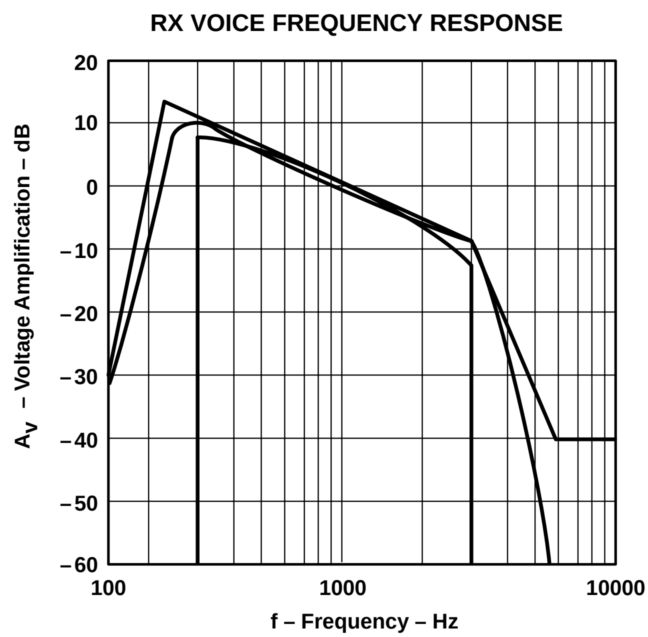


Figure 3

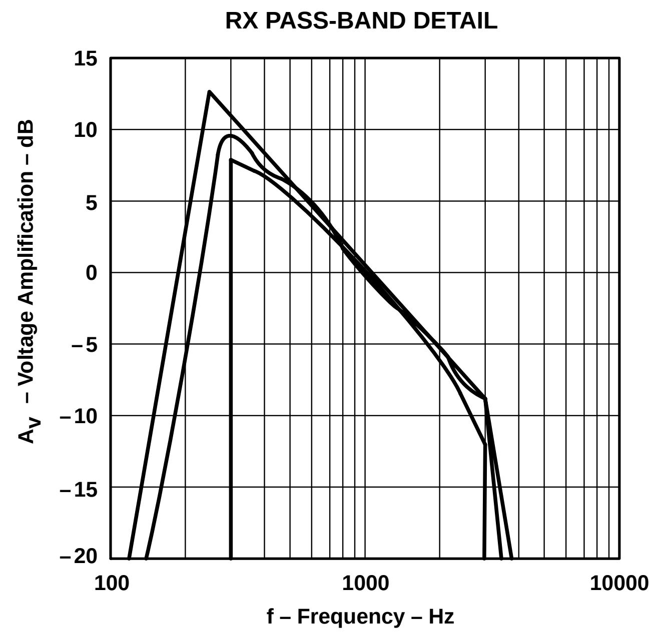


Figure 4

TYPICAL CHARACTERISTICS

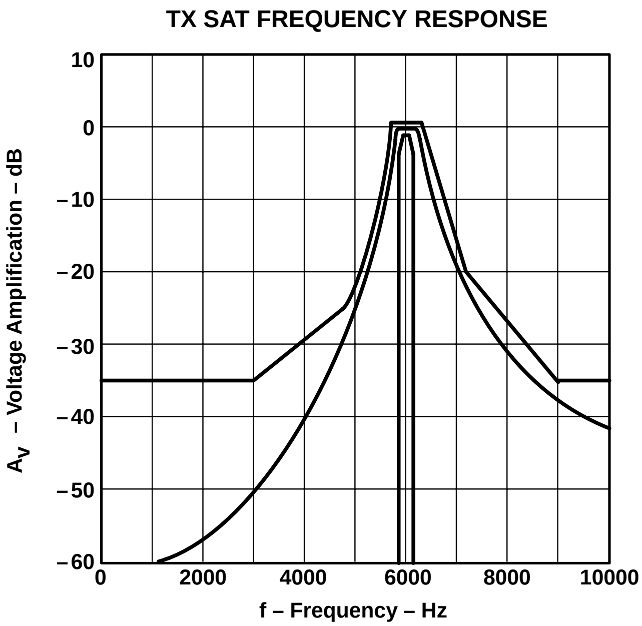


Figure 5

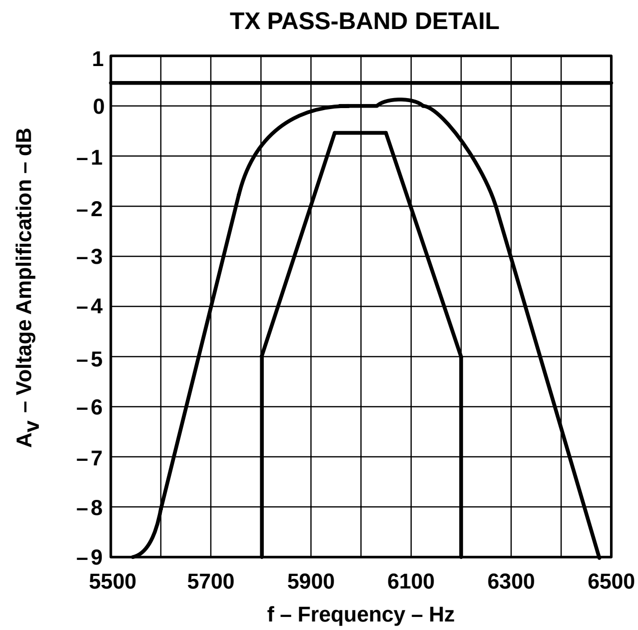


Figure 6

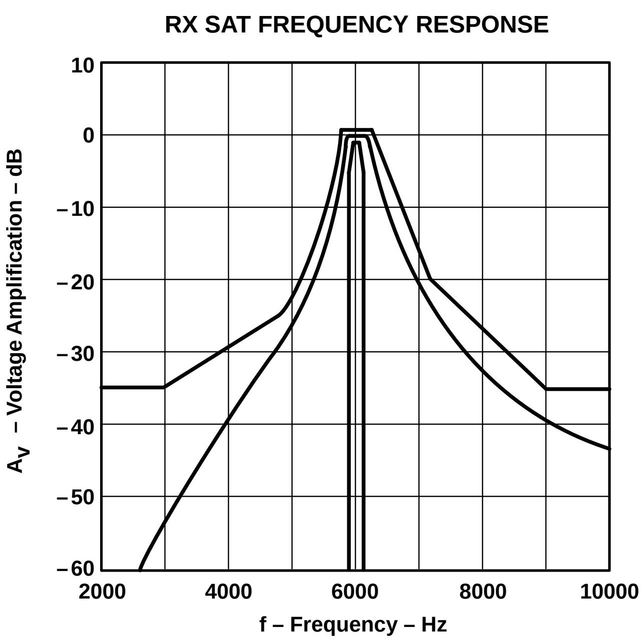


Figure 7

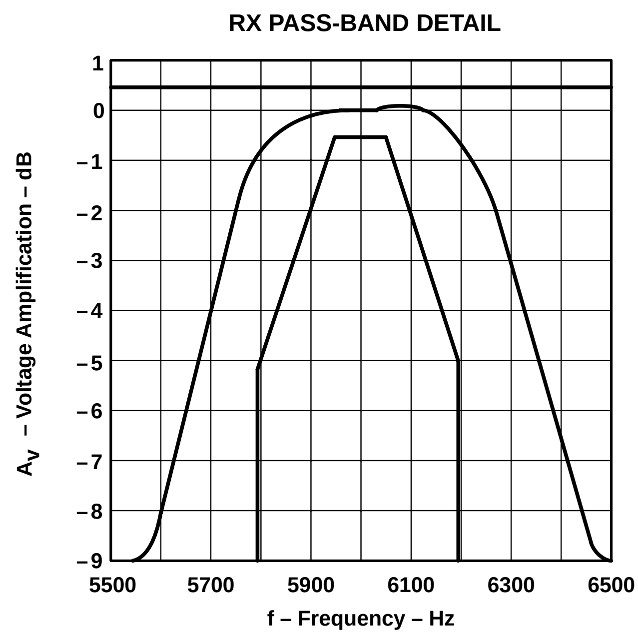


Figure 8

TYPICAL CHARACTERISTICS

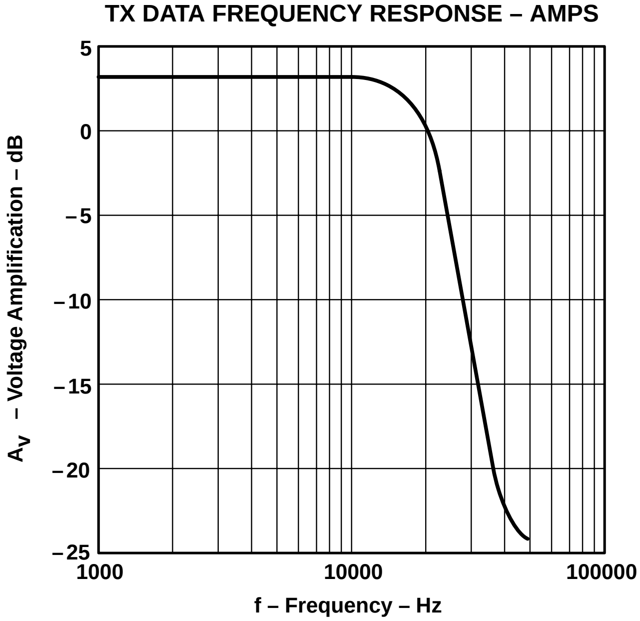


Figure 9

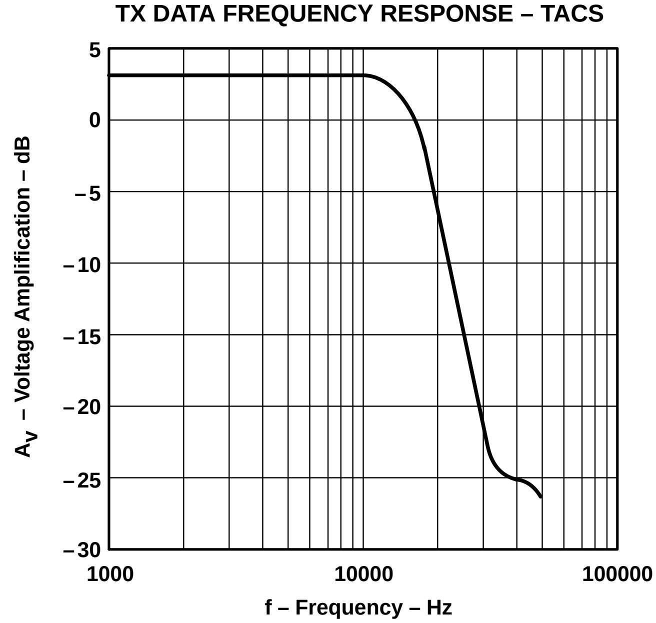


Figure 10

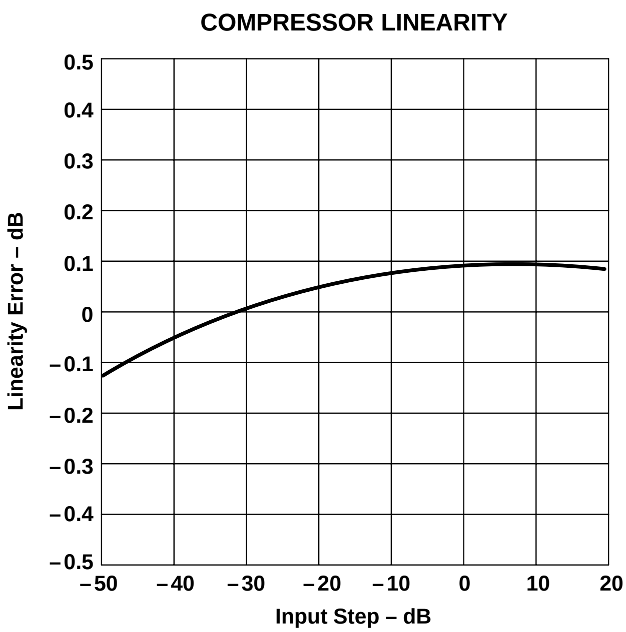


Figure 11

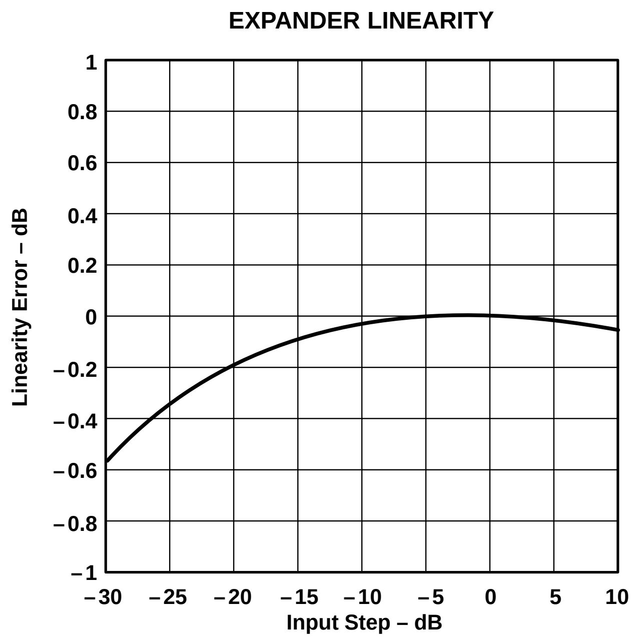


Figure 12

APPLICATION INFORMATION

analog cellular telephone baseband solution

The TCM8002 and TCM8010-50 chip set provides a complete solution to the audio and data filtering, decoding, and encoding required in a cellular telephone for the AMPS or TACS systems. The applications-circuit schematic is shown in Figure 13 and demonstrates that a minimum of external components is required.

The following extra functions are included in the TCM8010-50 and TCM8002:

- Three digital-to-analog converters
- Analog-to-digital converter
- Two timers
- I/O expansion

overall description

The following paragraphs detail the various function of the TCM8010-50 and TCM8002 chip set when used in this application.

TCM8010-50 transmit path

The inputs to the microphone amplifiers are MIC1 and MIC2. MIC1 could be used for the internal microphone and MIC2 for accessories (a hands-free unit). The TCM8010-50 is designed for single-supply operation. REF is provided to bias the noninverting inputs of the microphone amplifiers, M1P and M2P. The wideband data to be transmitted is input as a digital signal to TXDA. The TCM8010-50 then filters and provides a level trim for the signal.

The TCM8002 produces a digitally-filtered signal, phase locked to the received SAT. This is then connected to the input TXSA of the TCM8010-50, which filters and provides level adjustment for the digital signal. The output from the TCM8010-50 is at TXO and should be connected to the modulator in the RF section. The voice, wideband data, and SAT signal levels are programmable, eliminating the need for external adjustments.

TCM8010-50 receive path

The output from the FM demodulator/discriminator should be connected to the receive audio input (RXIN) of the TCM8010-50. Two audio outputs are provided at SP1 and SP2. These can be configured to be two separate outputs, with one driving the phone earpiece and the other for test or accessories (a hands-free unit) for example, or optionally can be configured to provide a differential output to increase the maximum level.

The TCM8010-50 filters and converts the received wideband data to a digital signal and outputs this at RXDO for connection to the TCM8002. The received SAT signal is filtered and converted to a digital signal. It is then made available at RXSO for transmission to the TCM8002.

TCM8010-50 digital-to-analog converters

Three uncommitted 8-bit DACs are included in the TCM8010-50 (DAC1OUT, DAC2OUT, and DAC3OUT). One can be used for power control of the RF transmit amplifier. The other two could be used to provide adjustment voltages for the RF stage such as calibrating the temperature-compensated crystal oscillator (TCXO) and trimming the first intermediate frequency (IF) stage.

TCM8010-50 analog-to-digital converter

Two multiplexed inputs to an ADC included in the TCM8010-50 are provided (ADCIN 1 and ADCIN 2). Possible uses are to measure battery voltage (using a potential divider) or received-signal-strength indicator (RSSI) voltage.

APPLICATION INFORMATION

TCM8002 transmit path

The data encoder includes all the necessary formatting for transmission on the control and voice channels. This digital signal is output at TXOUT. The received SAT digital signal is connected to TCM8002 SATIN and then the signal is recovered from the noise before being measured and regenerated. The digital output signal appears at TCM8002 SATOUT.

TCM8002 receive path

The received digital data signal is connected to RXIN for the control-and-voice channel data-recovery circuit. The data is then majority voted and error corrected. Finally, an interrupt is generated to signal to the microcontroller that there is received data available.

TCM8002 timers

A watchdog timer is provided that can reset the microcontroller in the telephone if a fault occurs. This is a requirement of both the AMPS and TACS systems.

An uncommitted programmable 8-bit timer is also available with an output labelled TMZERO that pulses low when the count reaches zero.

TCM8002 I/O expansion

Twenty programmable I/O lines are provided for the telephone microcontroller. These are individually bit-programmable as outputs or inputs with optional current source pullups.

An intelligent interface to the audio processor (TCM8010-50) provides an automatic audio-mute function when wideband data is being transmitted or received.

TCM8002 and TCM8010-50 clock and control

Both the TCM8002 and TCM8010-50 are connected to the microcontroller through the serial interface ($\overline{\text{CS}}$, DCLK, DATAIN, DATAOUT, INTERRUPT). The TCM8002 can be programmed to generate interrupts when events such as RX data available (data received) or the counter/timer reaching zero state occurs.

A low-power crystal oscillator is integrated into the TCM8002, and the CLKOUT output is provided for connection to the TCM8010-50.

APPLICATION INFORMATION

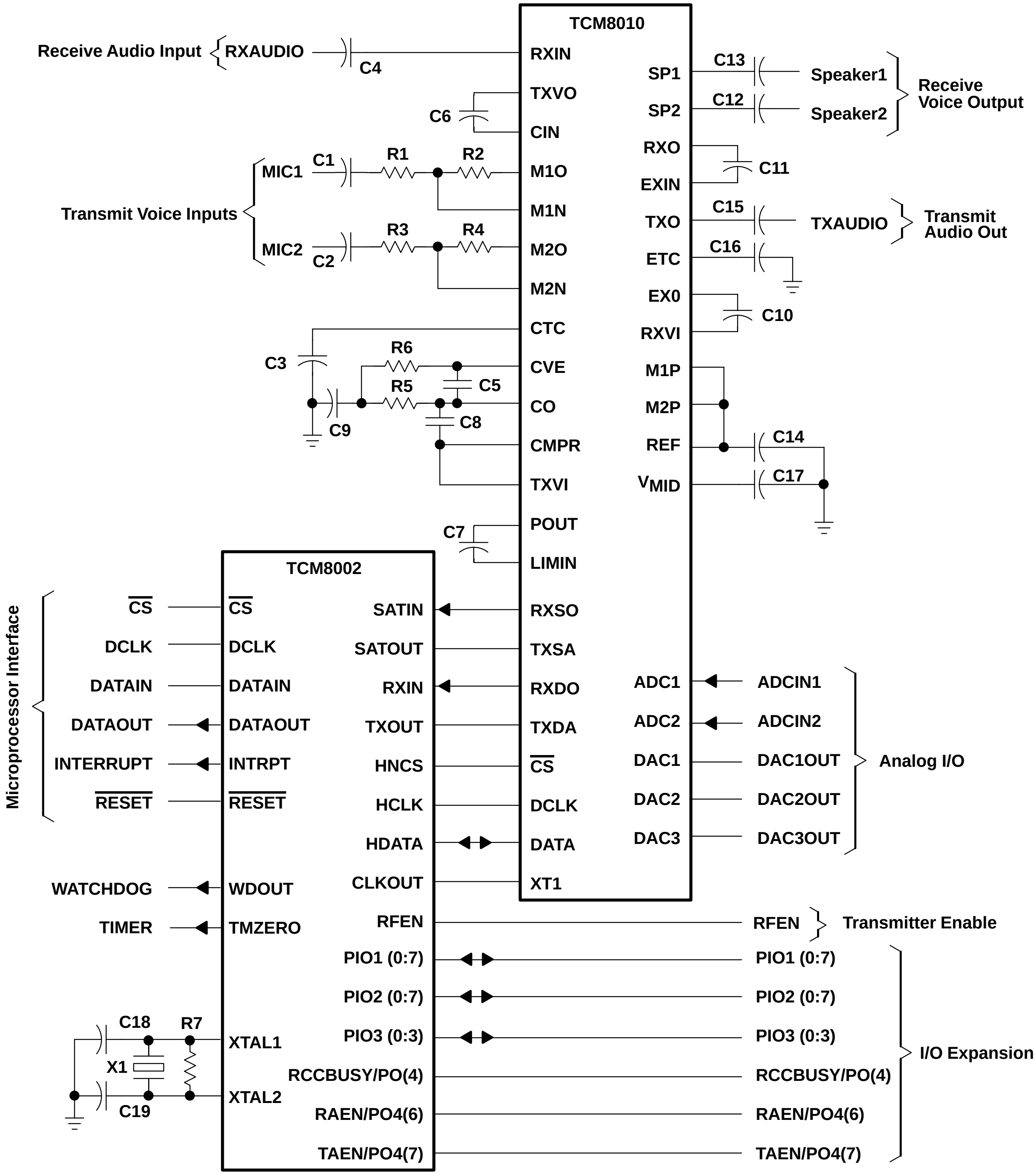


Figure 13. Complete Baseband Solution

APPLICATION INFORMATION

external component selection

COMPONENT DESIGNATION	TYPICAL VALUE	FUNCTION
R1	47 kΩ	Microphone preamplifier number 1 gain = R2/R1
R2	47 kΩ	Recommended minimum value
R3	47 kΩ	Microphone preamplifier number 2 gain = R4/R3
R4	47 kΩ	Recommended minimum value
R5	100 kΩ	Provides dc bias for the compressor
R6	100 kΩ	
R7	1 MΩ	Biasing resistor for crystal oscillator
C1	100 nF	AC couples the input to microphone preamplifier number 1 (MIC 1)
C2	100 nF	AC couples the input to microphone preamplifier number 2 (MIC 2)
C3	390 nF	Determines the attack and recovery times of the compressor
C4	10 nF	AC couples the receive audio and data input from the FM demodulator/discriminator
C5	47 pF	Required for HF stability of the compressor
C6	100 nF	AC couples the output of the selected microphone preamplifier to the compressor input. This is required because any dc offset would cause linearity errors.
C7	100 nF	AC couples the output of the preemphasis and bandpass filter to the limiter stage to ensure symmetrical clipping
C8	100 nF	AC couples the output of the compressor to the transmit switch (TXSW). Since this is also the compressor rectifier input, any dc offset would cause linearity errors.
C9	100 nF	AC decouples the compressor dc feedback
C10	100 nF	AC couples the output from the expander to the receive switch (RXSW)
C11	100 nF	AC couples the input to the expander to remove offsets that would otherwise cause linearity errors at low signal levels
C12	—	Required when the earpiece drive is single ended (not differential)
C13	—	
C14	470 nF	Decouples the resistor divider that produces REF, the input for the V_{MID} generator
C15	100 nF	AC couples the output from the transmit voice, data, and SAT signals to the FM modulator in the RF section
C16	330 nF	Determines the attack and recovery times of the expander
C17	470 nF	Provides a low ac impedance reference for the transmit and receive paths
C18	33 pF	Provides X1 with the required capacitive loading
C19	33 pF	Provides X1 with the required capacitive loading
X1	2.56 MHz	Crystal

printed circuit board layout precautions

Resistors R5 and R6 should be placed close to the TCM8010-50 to minimize stray capacitance between CO and CVE. Otherwise, compressor gain errors are caused at low signal levels and high frequencies.

suggested trim sequence

The TCM8010-50 and TCM8002 are designed so that no manual trims are required. All levels can be adjusted to meet the system requirements and compensate for production tolerances by writing to the digital interface. The data required can then be stored in a nonvolatile memory by the microcontroller in the telephone. When the telephone is turned on, an initialization routine can write this calibration data to the TCM8010-50.

APPLICATION INFORMATION

suggested trim sequence (continued)

The suggested sequence of adjustments for trimming is detailed below.

transmit

Step 1. TXTRIM

a. Set the TX data trim <DAT_2 – DAT_0> to nominal = <100>.

b. Set the TCM8002 and TCM8010-50 to transmit signaling tone.

c. Adjust <TXT_4 – TXT_0> to set the frequency deviation to that required by the system, AMPS or TACS.

Step 2. TXSAT

a. Turn the signaling tone off and turn on the SAT path. Input a 6-kHz signal to RXIN.

b. Adjust <SAT_3 – SAT_0> to give the required frequency deviation.

Step 3. MICTRIM

a. Mute the signaling tone and SAT.

b. Inject an audio signal at the desired level into the microphone preamplifier.

c. Adjust <$MICT_3$ – $MICT_0$> to set the frequency deviation.

Step 4. LIMITER TRIM

a. Increase the audio signal level by 20 dB typically.

b. Adjust <LIM_2 – LIM_0> to produce the required maximum deviation.

Step 5. DTMF TRIM

a. Mute the signaling tone, audio, and SAT.

b. For TACS, set bit <DTSK> to enable the skew of the levels between the low and high tones.

c. Turn on the DTMF generator and adjust <$DTTR_3$ – $DTTR_0$> to give the desired frequency deviation.

receive

Step 6. RXTRIM

Input a modulated signal to the telephone and adjust <RXT_3 – RXT_0> to produce the required level at SP1 and SP2.

RF stage

Step 7. DACs to trim RF section

Three 8-bit DACs can be used to trim sections of the RF stage using <$DACCAD_1$ – $DACAD_0$> to select the DAC and <DAC_7 – DAC_0> to set the level. <DACX2> sets the range of all three DACs and <DACON> enables all three outputs when the TCM8010-50 is in standby.

Typical uses would be RF transmit power control, TCXO trim, and first IF section trim.

The TCM8010-50 is designed for signal levels detailed in the following tables for AMPS and TACS systems. These tables suggest levels for both the transmitted and received audio, SAT, and DATA signals.

APPLICATION INFORMATION

transmit signal levels

AMPS mode

SIGNAL	PEAK FREQUENCY DEVIATION (kHz)	LEVEL AT TXO	UNIT
Design level	8	400	mVrms
Peak voice level	12	1697.1	mV peak to peak
SAT	2	100	mVrms
DATA	8	1131.4	mV peak to peak
DTMF low tone, 697 Hz	3.1365	156.8	mVrms
DTMF high tone, 1477 Hz	6.6465	332.3	mVrms

TACS mode

SIGNAL	PEAK FREQUENCY DEVIATION (kHz)	LEVEL AT TXO	UNIT
Design level	5.7	356.3	mVrms
Peak voice level	9.5	1697.6	mV peak to peak
SAT	1.7	106.3	mVrms
DATA	6.4	1131.5	mV peak to peak
DTMF low tone, 697 Hz	1.2 max	75	mVrms
DTMF high tone, 1477 Hz	3.19 max	199.4	mVrms

receive signal levels

AMPS mode

SIGNAL	PEAK FREQUENCY DEVIATION (kHz)	LEVEL AT RXIN	UNIT
Design level	8	400	mVrms
Peak voice level	12	1697.1	mV peak to peak
SAT	2	100	mVrms
DATA	8	1131.4	mV peak to peak

TACS mode

SIGNAL	PEAK FREQUENCY DEVIATION (kHz)	LEVEL AT RXIN	UNIT
Design level	5.7	356.3	mVrms
Peak voice level	9.5	1697.6	mV peak to peak
SAT	1.7	106.3	mVrms
DATA	6.4	1131.5	mV peak to peak

PRINCIPLES OF OPERATION

general

The TCM8010-50 consists of a number of functional blocks and is controlled by the digital interface. The control bits associated with each block are shown in the angled brackets symbol < >. In standby mode <STBY>, the receive data path from RXIN to RXDO is on and the DACs can be on or off as required. All other parts of the device, including the crystal oscillator, are off. When in the active mode, the receive and transmit paths and the DAC blocks are continuously on, and the DTMF and ADC blocks are turned on as required.

Control bits <MD_1 – MD_0> set the TCM8010-50 for the desired system (AMPS or TACS).

transmit path

The transmit path on the TCM8010-50 consists of a number of functional blocks, which are described in the following paragraphs.

mic inputs

Voice signals are input via a pair of microphone preamplifiers, which are stable for gains between 0 dB and 20 dB. All voice-path specifications are given with the preamplifiers configured as unity-gain inverting amplifiers. In standby mode, the bias to the microphone preamplifiers is turned off and the outputs M1O and M2O are in the high-impedance state.

MICSW

The MICSW block is a 2-input switch that selects either of the preamplifier outputs, and is under control of the digital interface (<MICSEL>).

MICTRIM

The MICTRIM block provides gain adjustment to compensate for differing microphone sensitivities (<$MICT_3$ – $MICT_0$>). A second-order Sallen-Key low-pass filter is incorporated in this block to provide antialiasing for the TX voice signal.

compressor

The compressor provides a 1-dB change in output signal level for a 2-dB change in input level over an operating input range of 50 dB. The unity-gain point, $V_{ref,}$ is proportional to the value of V_{DD} (see the compressor table in the transmit path electrical charactistics). Attack time is measured by increasing the input-signal amplitude by a 12-dB step relative to 13 mV rms and is defined as the time required for the output envelope to reach 1.5 times the final steady-state level. Recovery time is measured by reducing the input signal amplitude by a 12-dB step to 13 mV rms and is defined as the time required for the output envelope to settle to 0.75 times the final steady-state level.

The attack and recovery times are determined by an internal resistor (R_{COMP}) and the external capacitor, C_{CTC}, connected between CTC and 0 V, V_{SS}.

Attack time = $0.151 \times C_{CTC} \times R_{COMP}$

Recovery time = $0.693 \times C_{CTC} \times R_{COMP}$

TXSW

This block is a 3-input switch that selects either the compressor output, compressor bypass (for testing), or the output of the DTMF generator. TXSW is controlled by <$TXSW_1$ – $TXSW_0$>.

PRINCIPLES OF OPERATION

TXATTEN

The output of TXSW passes through the TXATTEN block, which provides four levels of attenuation.

preemphasis

The output from TXATTEN is connected to the preemphasis block, which provides the necessary 6 dB per octave increase in gain with frequency by using a second-order filter. Also included in this block is an eighth-order band-pass filter function with a 300-Hz to 3-kHz passband. The nominal gain of this stage is 6 dB at 1 kHz and its output is routed to the POUT terminal.

limiter

The limiter block limits the maximum output under overload signal conditions, and the limit level is adjustable under control of the serial interface <LIM_2 – LIM_0>. The limiter range is designed to allow the TX path distortion and maximum signal output specifications to be achieved at a single limiter-adjustment code. The output of the preemphasis block is ac coupled (via an external capacitor) into the LIMIN terminal to ensure symmetrical limiting.

low-pass filter

The limiter output is processed by the low-pass filter block, which is a fourth-order low-pass filter plus second-order equalizer, to remove excessive harmonics produced by the limiting process.

TXSUM

This block can sum together or mute any of its three inputs (SAT, data, and voice) under the control of the <TXSAT, TXDAT, TXVOX> bits, respectively.

TXTRIM

The TXTRIM gain-adjust block can be used to compensate for different modulator sensitivities using bits <TXT_4 – TXT_0>. A continuous-time output low-pass smoothing filter is included with a typical cutoff frequency of 30 kHz.

TX data filter

Transmit data is input to terminal TXDA and is routed to the TX data filter block where the data is first conditioned by a second-order antialiasing filter before going on to the transmit data filter.

In the AMPS and TACS modes, the transmit data is a Manchester-encoded digital signal at 10k bit/s for AMPS or 8k bits/s for TACS. The transmit data filter for these two modes is a fourth-order Butterworth low-pass filter, with its – 3-dB point switchable between AMPS and TACS modes.

The filtered AMPS or TACS wideband-data signal is summed into the transmit signal path in the TXSUM block.

TXSAT filter path – TXSA to TXO

The input to the transmit SAT signal path is determined by the SATSW block, which selects between the TXSA terminal and the output of the receive SAT filter (RXSAT). The signal is processed by the TXSAT filter block, which includes an antialiasing filter, a fourth-order narrow-band band-pass filter centered at 6 kHz, and a gain adjust stage <SAT_3 – SAT_0>. The output of this block is then applied to an input of TXSUM to be summed into the voice path when selected.

PRINCIPLES OF OPERATION

receive path – voice

The demodulated signal from the receiver is input to the TCM8010-50 at the RXIN terminal. A pair of loudspeaker drivers are provided, producing output signals on terminals SP1 and SP2, and are capable of driving 500-Ω loads.

RXTRIM

A gain-adjust block, RXTRIM, is provided to allow variations in the receiver FM demodulator/discriminator characteristics to be accommodated <RXT_3 – RXT_0>. This block is enabled in both active and standby modes. A second-order continuous-time filter with a typical cutoff frequency of 30 kHz provides an antialiasing function for the receive signal path.

deemphasis

The deemphasis filter block exhibits a 6-dB/octave decrease in gain versus frequency characteristic. It also includes an eighth-order band-pass filter (pass band = 300 Hz to 3 kHz) to separate the received voice signal from the data and SAT signals. A continuous-time smoothing filter is incorporated at the output, and the output signal appears at terminal RXO.

expander

The expander block provides a 2-dB change in output signal level for a 1-dB change in input level over an operating input range of 33 dB. The unity-gain level, V_{ref}, is proportional to V_{DD} (see the expander table in the receiver path electrical charactistics). Attack time is measured by increasing the input signal amplitude by a 6-dB step relative to 72.5 mV and is defined as the time required for the output envelope to reach 0.57 times the final steady-state level. Recovery time is measured by reducing the input signal amplitude by a 6-dB step to 72.5 mV and is defined as the time required for the output envelope to settle to 1.5 times the final steady-state level.

The attack and recovery times are determined by an internal resistor, R_{EXP}, and the external capacitor, C_{EXP}, connected to ETC and 0 V, V_{SS}.

Attack time = $0.173 \times C_{ETC} \times R_{EXP}$

Recovery time = $0.693 \times C_{ETC} \times R_{EXP}$

RXSW

RXSW is a 4-input switch block that provides a selection between the call-tone input terminal (CTI), the expander output (externally capacitively coupled to terminal RXVI), the expander-bypass path (for testing), and the output from the DTMF generator as the input to the volume-control block. The control bits are <$RXSW_1$ and $RXSW_0$>.

To simplify the connection of a digital signal for a *User Alert* tone (typically between 200 Hz and 400 Hz), no internal bias is provided for the CTI input. If an ac-coupled signal is applied to CTI, an external bias resistor (typical value is 100 kΩ) is required and should be connected between CTI and V_{MID}.

volume control

This block provides output level adjustment to implement a user-adjustable level control via control bits <VOL_3 – VOL_0>.

LSSW

The loudspeaker control switch block (LSSW) allows selection between either SP1 or SP2 outputs, muting, or differential drive of both terminals via control bits <LS_1 – LS_0>.

PRINCIPLES OF OPERATION

receive path – data/SAT

The demodulated signal from the receiver is input to the TCM8010-50 at the RXIN terminal. This signal, containing voice, data, and SAT components, is processed and antialias filtered by the RXTRIM block.

receive data path – data comparator

The signal from RXTRIM is applied to the data comparator block, which has defined threshold levels. The data signal is Manchester encoded at 10 kbit/s for AMPS mode and at 8 kbit/s for TACS mode. Detected data appears at RXDO. This signal path is enabled in the standby mode.

receive SAT path – RX SAT filter

The RX SAT filter block uses a fourth-order Butterworth bandpass filter centered at 6 kHz to separate received SAT signals from the voice signal. The output of the bandpass filter is routed to an input of the SATSW block and to the SAT comparator block.

receive SAT path – SATSW

SATSW is a 2-input switch block that selects between the output of the RX SAT filter and an external SAT source (applied to terminal TXSA) via control bit <ISAT>.

receive SAT path – SAT comparator

The SAT comparator block recovers the SAT signal and has defined hysteresis levels for improved noise immunity. The output is routed to terminal RXSO. An internal switch, controlled by bit <SATDIG>, bypasses the SAT comparator, applying the output from the RX SAT filter block directly to terminal RXSO.

digital interface

The TCM8010-50 is controlled by a 3-wire digital interface, consisting of a clock signal (DCLK), a chip select ($\overline{CS}$), and a bidirectional data line (DATA). The logic signal present on DATA is written into the device on the rising edge of DCLK when $\overline{CS}$ is low. Serial messages to and from the device contain a read/write bit, an address field, and a data word. Results from the ADC are read back using the serial interface, and the DCLK signal is used to drive the converter. Test access to analog and digital sections of the device are provided using the serial interface.

write operations

A timing diagram for a write operation to the device is shown in Figure 14. In this case, the read/write bit is set to 1, followed by a 3-bit address word, (A2–A0), and a 10-bit data word (D9–D0). Data shifts into the device on the rising edge of DCLK and is transferred to internal registers on the falling edge of the fourteenth clock pulse after $\overline{CS}$ goes low. If $\overline{CS}$ returns high before this time, no transfer takes place and the input interface is reset.

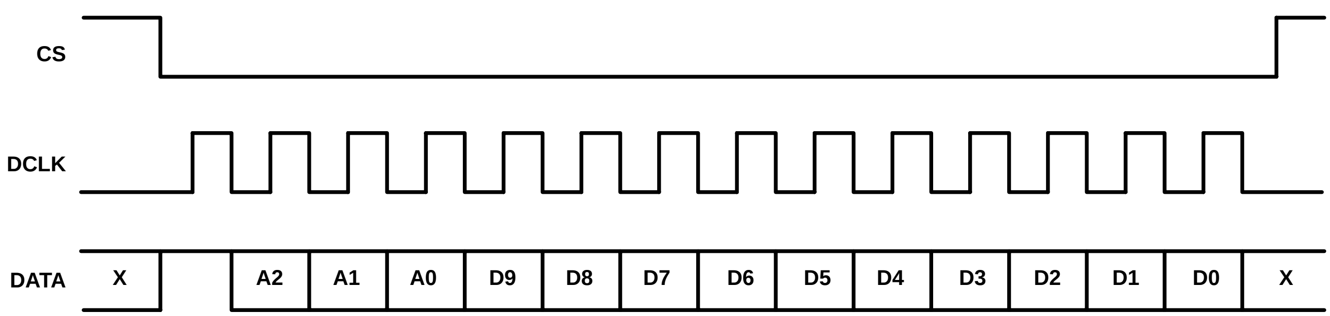


Figure 14. Write-Operation Timing

PRINCIPLES OF OPERATION

control words

Table 1 shows control-word and configuration assignments for the device. Table 2 shows control-word descriptions, Table 3 shows test modes, and Table 4 details DTMF control words.

Table 1. Control-Word and Configuration Assignments

	A_2	A_1	A_0	D_9	D_8	D_7	D_6	D_5	D_4	D_3	D_2	D_1	D_0
Word 0	0	0	0	STBY	MD_1	MD_0	ISAT	SATDIG	DACX2	CKSEL	$CKRT_2$	$CKRT_1$	$CKRT_0$
Word 1	0	0	1	$TXSW_1$	$TXSW_0$	TXSAT	TXDAT	TXVOX	$TXATT_1$	$TXATT_0$	DACON	LS_1	LS_0
Word 2	0	1	0	MICSEL	$MICT_3$	$MICT_2$	$MICT_1$	$MICT_0$	TXT_4	TXT_3	TXT_2	TXT_1	TXT_0
Word 3	0	1	1	LIM_2	LIM_1	LIM_0	SAT_3	SAT_2	SAT_1	SAT_0	1	0	0
Word 4	1	0	0	RXT_3	RXT_2	RXT_1	RXT_0	$RXSW_1$	$RXSW_0$	VOL_3	VOL_2	VOL_1	VOL_0
Word 5	1	0	1	DTSK	$DTTR_3$	$DTTR_2$	$DTTR_1$	$DTTR_0$	0	0	0	$TEST_1$	$TEST_0$
Word 6	1	1	0	$DACAD_1$	$DACAD_0$	DAC_7	DAC_6	DAC_5	DAC_4	DAC_3	DAC_2	DAC_1	DAC_0
Word 7	1	1	1	—	—	—	—	—	—	$DTMF_3$	$DTMF_2$	$DTMF_1$	$DTMF_0$

Table 2. Control-Word Descriptions

	DESCRIPTION
Word 0	STBY = Standby select: 0 = Standby, 1 = Active MD_1 – MD_0 = Mode select: 00 = AMPS, 01 = Undefined, 10 = TACS, 11 = Undefined ISAT = SAT select: 0 = External, 1 = Internal SATDIG = Digital/Analog RX SAT: 0 = Digital, 1 = Analog DACX2 = DAC range select: 0 = 0 – V_{DD}/2, 1 = 0 – V_{DD} CKSEL = Clock source select: 0 = Oscillator, 1 = Sinusoidal input $CKRT_2$ – $CKRT_0$ = Clock rate select: 000 = 3.58 MHz, 001 = 7.16 MHz, 010 = 10.74 MHz, 011 = 14.32 MHz, 100 = 2.56 MHz, 101 = 10.24 MHz, 110 = 12.80 MHz, 111 = 15.36 MHz
Word 1	$TXSW_1$ – $TXSW_0$ = TX Voice select: 00 = Mute, 01 = Compressor O/P, 10 = Compressor bypass, 11 = DTMF TXSAT = TX SAT enable: 0 = Mute, 1 = Enable TXDAT = TX Wideband data enable: 0 = Mute, 1 = Enable TXVOX = TX Voice enable: 0 = Mute, 1 = Enable $TXATT_1$ – $TXATT_0$ = TX attenuation: 00 = 0 dB, 01 = 8 dB, 10 = dB, 11 = 24 dB DACON = DACS on select in standby: 0 = Off, 1 = On LS_1 – LS_0 = Loudspeaker configuration: 00 = Mute, 01 = SP2 enable, 10 = SP1 enable, 11 = Differential
Word 2	MICSEL = Microphone select: 0 = M1, 1 = M2 $MICT_3$ – $MICT_0$ = Microphone trim: 0000 = minimum gain, 1111 = maximum gain TXT_4 – TXT_0 = TX Deviation trim: 00000 = minimum gain, 11111 = maximum gain
Word 3	LIM_2 – LIM_0 = Deviation limiter adjust: 000 = minimum deviation, 111 = maximum deviation SAT_3 – SAT_0 = TXSAT adjust: 0000 = minimum, 1111 = maximum D_0 – D_1 = 0, D_2 = 1
Word 4	RXT_3 – RXT_0 = RX input adjust: 0000 = minimum, 1111 = maximum $RXSW_1$ – $RXSW_0$ = RX switch control: 00 = CT input, 01 = Expander O/P, 10 = Expander bypass, 11 = DTMF VOL_3 – VOL_0 = RX path audio volume control: 0000 = minimum, 1111 = maximum
Word 5	DTSK = DTMF Skew enable: 0 = disabled, 1 = enabled $DTTR_3$ – $DTTR_0$ = DTMF adjust: 0000 = minimum, 1111 = maximum $TEST_1$ – $TEST_0$ = Test mode: (see Table 3) D_2 – D_4 = 0
Word 6	$DACAD_1$ – $DACAD_0$ = DAC address: 00 = DAC 1, 01 = DAC 2, 10 = DAC 3, 11 = all DACs
Word 7	$DTMF_3$ – $DTMF_0$ = DTMF control: (see Table 4)

PRINCIPLES OF OPERATION

Table 3. Test Modes (to Word 5)

SATDIG	$TEST_1$	$TEST_0$	MODE	OUTPUT AT RXDO	OUTPUT AT RXSO
X	0	0	Normal		
0	0	1	Digital Test	Digital DTMF High Tone	Digital DTMF Low Tone
1	1	0	Analog Test A	Rx Data (analog)	Limiter Output (dc)
1	1	1	Analog Test B	Bandgap Output	Tx Data (analog)
0	1	0	Other States	Rx Data (digital)	Rx SAT (digital)
0	1	1	Other States	Digital DTMF High Tone	Digital DTMF Low Tone
1	0	1	Other States	Digital DTMF High Tone	RXSAT (analog)

Table 4. DTMF Control (to Word 7)

$DTMF_3$	$DTMF_2$	$DTMF_1$	$DTMF_0$	KEY	LOW TONE Hz	HIGH TONE Hz
0	0	0	0	1	697	1209
0	0	0	1	4	770	1209
0	0	1	0	7	852	1209
0	0	1	1	*	941	1209
0	1	0	0	2	697	1336
0	1	0	1	5	770	1336
0	1	1	0	8	852	1336
0	1	1	1	0	941	1336
1	0	0	0	3	697	1477
1	0	0	1	6	770	1477
1	0	1	0	9	852	1477
1	0	1	1	#	941	1477
1	1	0	0	—	697	Off
1	1	0	1	—	Off	1209
1	1	1	0	—	Off	1477
1	1	1	1	—	Off	Off

PRINCIPLES OF OPERATION

read operations

A timing diagram of a read operation, which outputs ADC results from the device, is shown in Figure 15. The first bit driven into the device is a logic 0, followed by a 3-bit address word. The device then assumes control of the DATA line on the falling edge of the fifth clock pulse after $\overline{CS}$ goes low. The conversion result is output MSB first, with the MSB being output on the falling edge of the seventh clock pulse after $\overline{CS}$ goes low. Control of the DATA line is released (returned to input mode), when $\overline{CS}$ goes high.

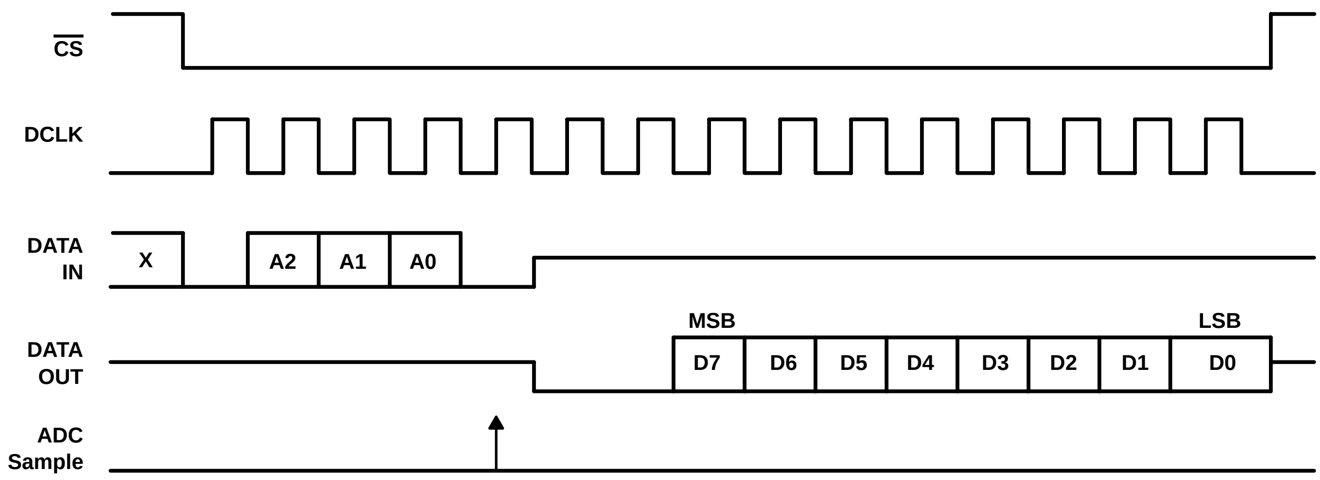


Figure 15. Read-Operation Timing

Table 5 details the decoding of the three address bits.

Table 5. Address Bit Decoding

A2	A1	A0	REFERENCE	MEASUREMENT
0	0	0	Band gap	V_{MID}
0	0	1	Band gap	ADC1
0	1	0	Band gap	ADC2

additional functions

The following paragraphs detail some additional functions of the TCM8010-50.

digital-to-analog converters

Three 8-bit, voltage-output DACs are provided, with outputs on terminals DAC1, DAC2, and DAC3. The output range of each converter is from 0 V to $V_{DD}/2$ or 0 V to V_{DD} with an LSB step size of $V_{DD}/256$ or $V_{DD}/2 \times 1/256$ as selected by <DACX2>. All DAC outputs can either go to 0 V in standby mode or be active on depending on the state of control bit <DACON>. For correct operation of all of the TCM8010-50, <DACON> must be set to 0 in active mode. Previously written values are restored to the DAC outputs on entry to active mode. <$DACAD_1$–$DACAD_0$> selects which DAC is being addressed, and <DAC_7–DAC_0> sets the output voltage.

PRINCIPLES OF OPERATION

analog-to-digital converter

The TCM8010-50 contains an 8-bit ADC with a 3-channel analog-input multiplexer. This allows conversion of signals on ADC1, ADC2, and V_{MID}. An internal band gap voltage reference multiplied by two is used when measuring ADC1, ADC2, and V_{MID}.

Fifteen periods of DCLK are required to complete a conversion.

DTMF generator

The DTMF generator produces the seven standard tones with a frequency accuracy of ±1%. The desired DTMF signal is selected by <$DTMF_3$–$DTMF_0$>. A switchable preemphasis or skew between the low and high tone groups is provided for TACS operation and is selected by bit <DTSK>.

DTMF signal levels scale directly with supply voltage. A 4-bit trim is provided to allow adjustment of DTMF amplitude to meet system specifications and allow flexibility for user-generated call-tone type signals (<$DTTR_3$–$DTTR_0$>). When DTMF is selected in the transmit or receive paths, typical voice signals are attenuated by 50 dB.

clock and supply

Power supply and clock considerations are covered in the following paragraphs.

supply voltage

Specifications are given for a supply voltage of 5 V. Signal levels such as SAT, DATA, and DTMF are derived from this. Other parameters such as the compressor and expander unity-gain levels are also dependent on the supply voltage.

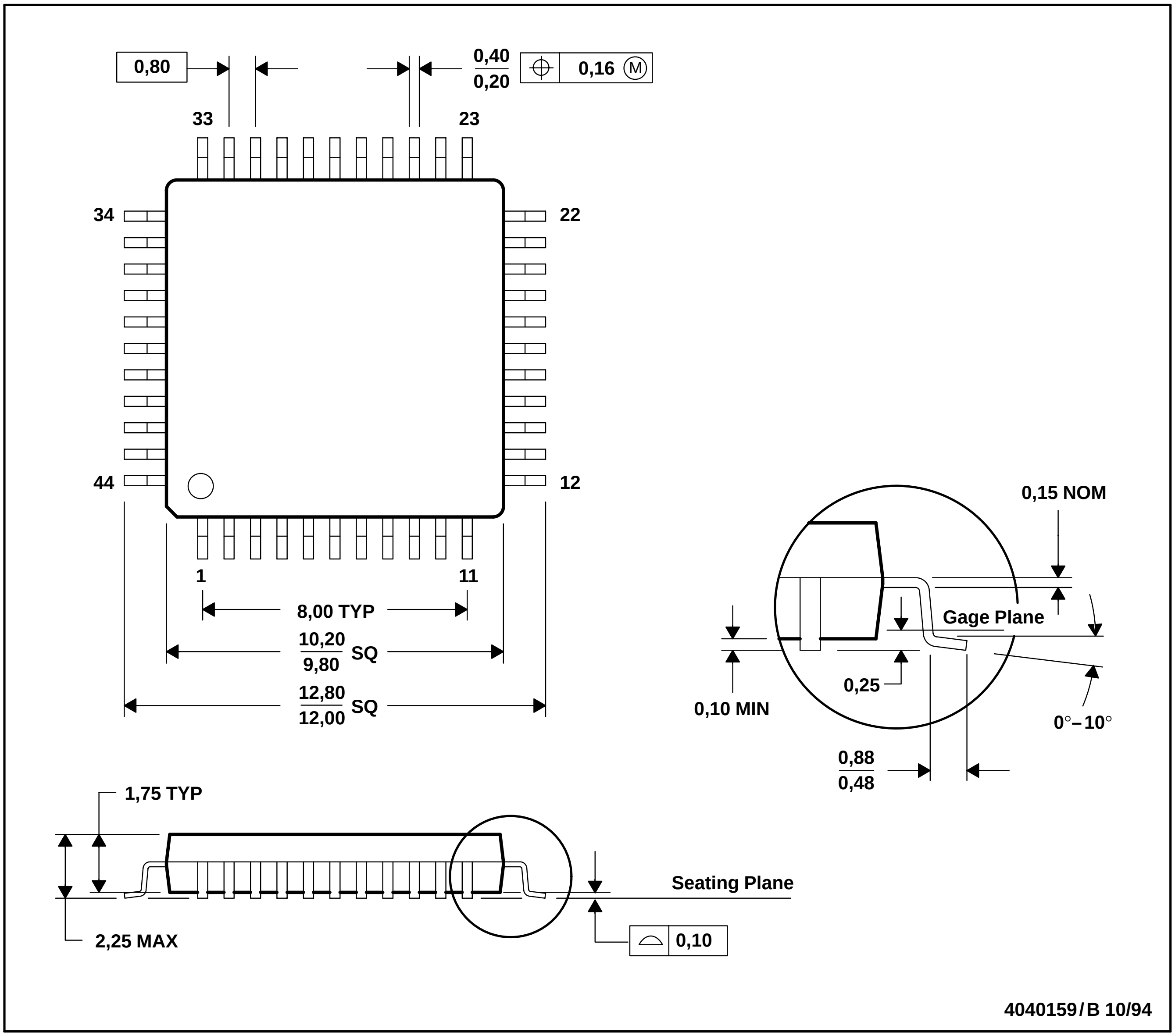
supply current

The TCM8010-50 has two basic operating modes: standby and active. In the standby mode, only the receive data path is enabled and current consumption is less than 2 mA. There is also the option of keeping the DACs powered up in the standby mode, depending on the setting of <DACON>. In the active mode, all functional blocks are powered up and the current consumption is less than 12 mA.

crystal oscillator and clock interface

The clock signal for the device can be generated by the internal oscillator block using an external crystal connected to the XT0 and XT1 terminals. Or, an external 0.5-V (minimum) peak sinusoidal clock signal can be applied to XT1. The external clock signal or the crystal can be one of eight frequencies, selected by control bits <$CKRT_2$–$CKRT_0$>. Crystal or external clock operation is selected by <CKSEL>.

FR (S-PDFP-G44) **PLASTIC QUAD FLATPACK**



NOTES: A. All linear dimensions are in millimeters.
B. This drawing is subject to change without notice.

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