

DATA SHEET

TDA9615H

Audio processor for VHS hi-fi

Preliminary specification
File under Integrated Circuits, IC02

1997 Jun 16

Audio processor for VHS hi-fi**TDA9615H****FEATURES**

- All functions controlled via the 2-wire I²C-bus
- Single supply
- Integrated standby modes for low power consumption
- Integrated power muting for line and RFC output
- Full support of video recorder feature modes
- Audio level meter output
- Hi-fi signal processing:
 - Adjustment-free
 - High performance
 - Patented low distortion switching noise suppressor
 - NTSC and PAL (SECAM) standard
- Linear audio input with level adjustment
- 5 stereo inputs and additional mono audio input
- 2 stereo outputs (line and decoder) with independent output selection
- RF converter output with overload protect AGC.

**GENERAL DESCRIPTION**

The TDA9615H is an audio control and processing circuit for VHS hi-fi video recorders, controlled via the I²C-bus. The device is adjustment-free using an integrated auto-calibration circuit. Extensive input and output selection is offered, including full support for (Euro-SCART) pay-TV decoding and video recorder feature modes.

ORDERING INFORMATION

TYPE NUMBER	PACKAGE		
	NAME	DESCRIPTION	VERSION
TDA9615H	QFP44	plastic quad flat package; 44 leads (lead length 1.3 mm); body 10 × 10 × 1.75 mm	SOT307-2

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BLOCK DIAGRAM

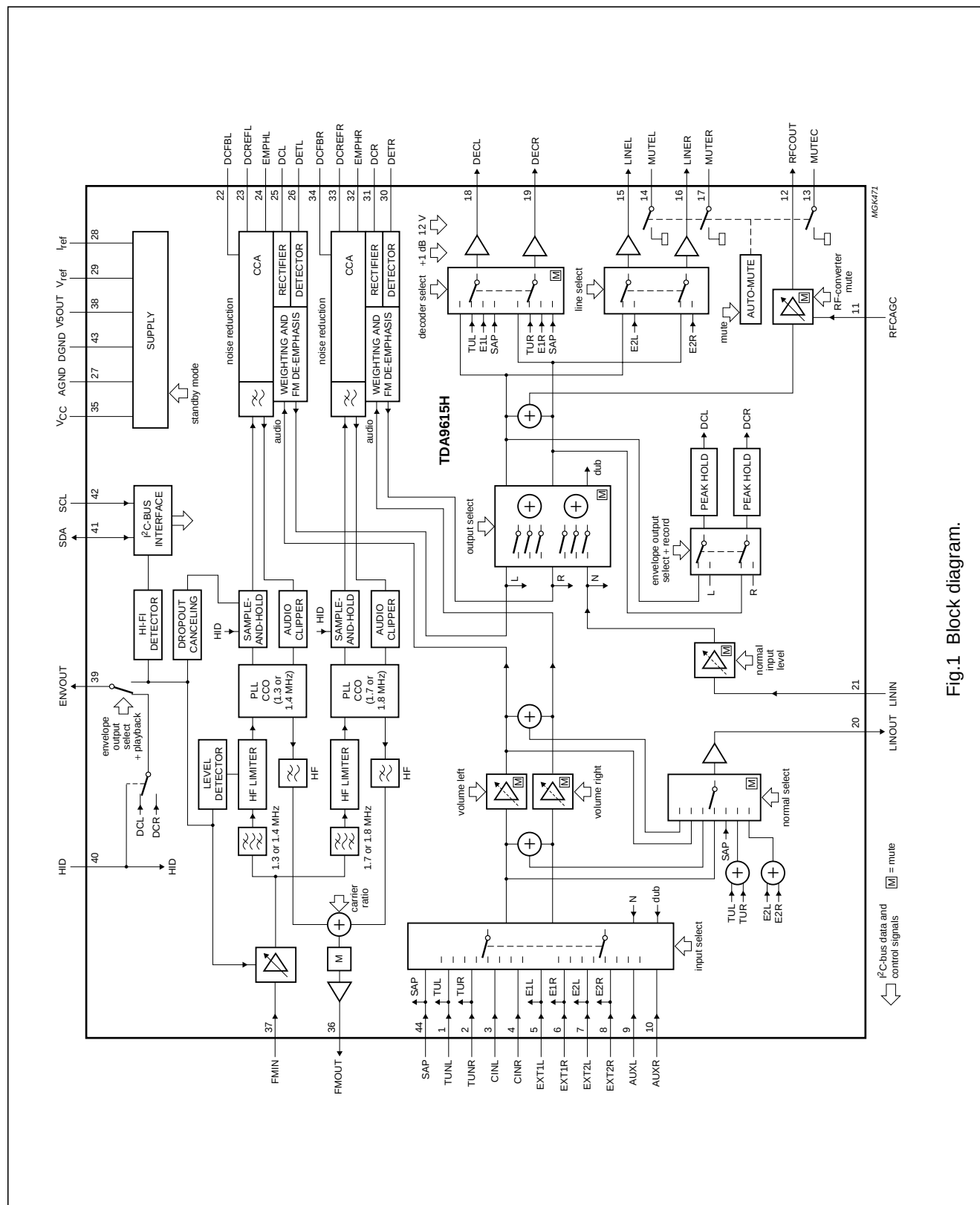


Fig.1 Block diagram.

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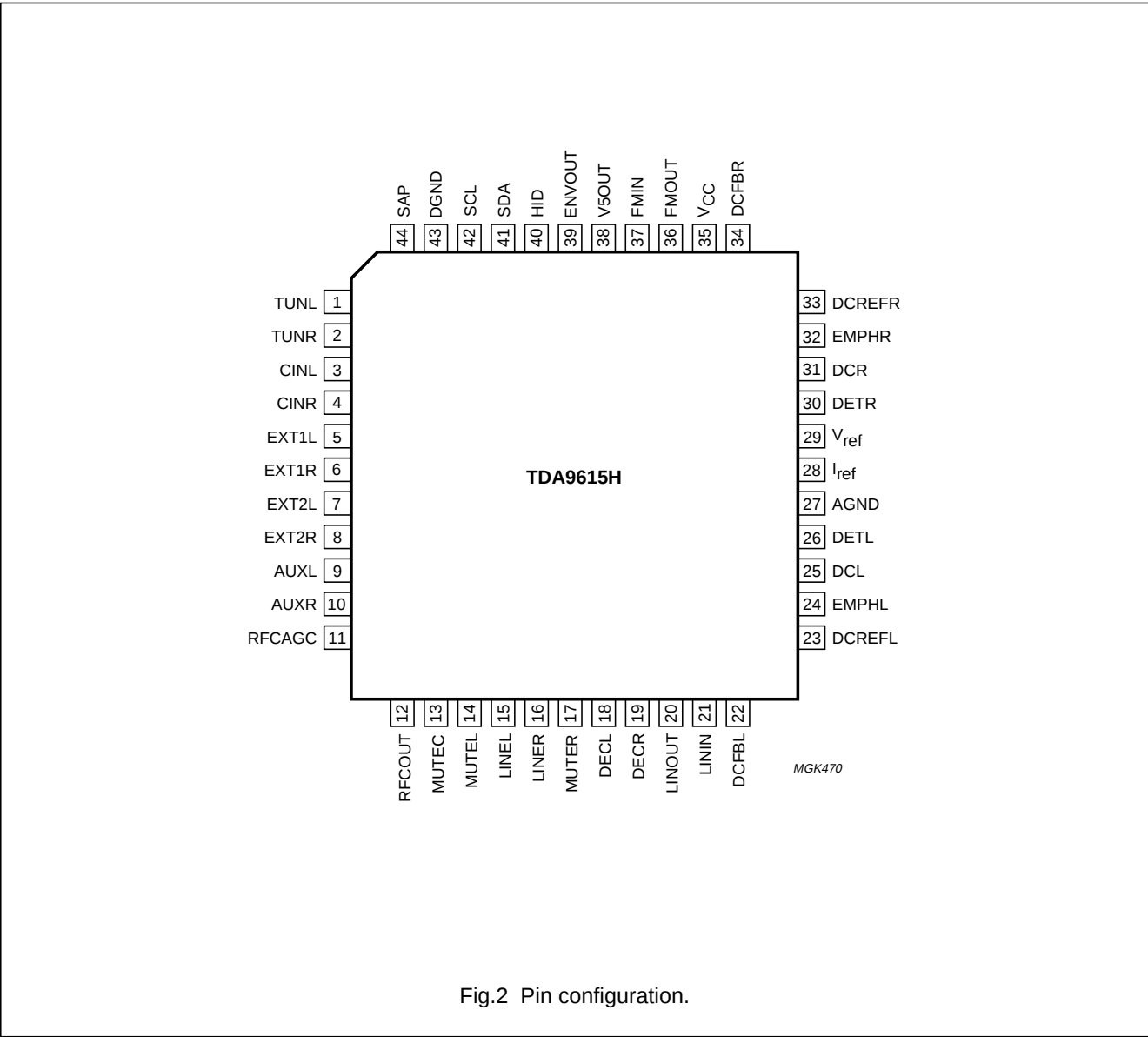
PINNING

SYMBOL	PIN	DESCRIPTION
TUNL	1	tuner input left
TUNR	2	tuner input right
CINL	3	cinch input left
CINR	4	cinch input right
EXT1L	5	external 1 input left
EXT1R	6	external 1 input right
EXT2L	7	external 2 input left
EXT2R	8	external 2 input right
AUXL	9	auxiliary input left
AUXR	10	auxiliary input right
RFCAGC	11	RFC AGC timing input
RFCOUT	12	RFC output
MUTEC	13	mute for RFC output
MUTEL	14	mute for line output left
LINEL	15	line output left
LINER	16	line output right
MUTER	17	mute for line output right
DECL	18	decoder output left
DECR	19	decoder output right
LINOUT	20	linear audio output
LININ	21	linear audio input
DCFBL	22	NR DC feedback left

SYMBOL	PIN	DESCRIPTION
DCREFL	23	NR DC reference left
EMPHL	24	NR emphasis left
DCL	25	NR DC decoupling left
DETL	26	NR detector left
AGND	27	analog ground
I_{ref}	28	reference current standard
V_{ref}	29	reference voltage filter
DETR	30	NR detector right
DCR	31	NR DC decoupling right
EMPHR	32	NR emphasis right
DCREFR	33	NR DC reference right
DCFBR	34	NR DC feedback right
V_{CC}	35	supply voltage
FMOOUT	36	FM output
FMIN	37	FM input
V5OUT	38	5 V decoupling output
ENVOUT	39	envelope output
HID	40	HID input
SDA	41	I ² C-bus SDA input/output
SCL	42	I ² C-bus SCL input
DGND	43	digital ground
SAP	44	tuner SAP input

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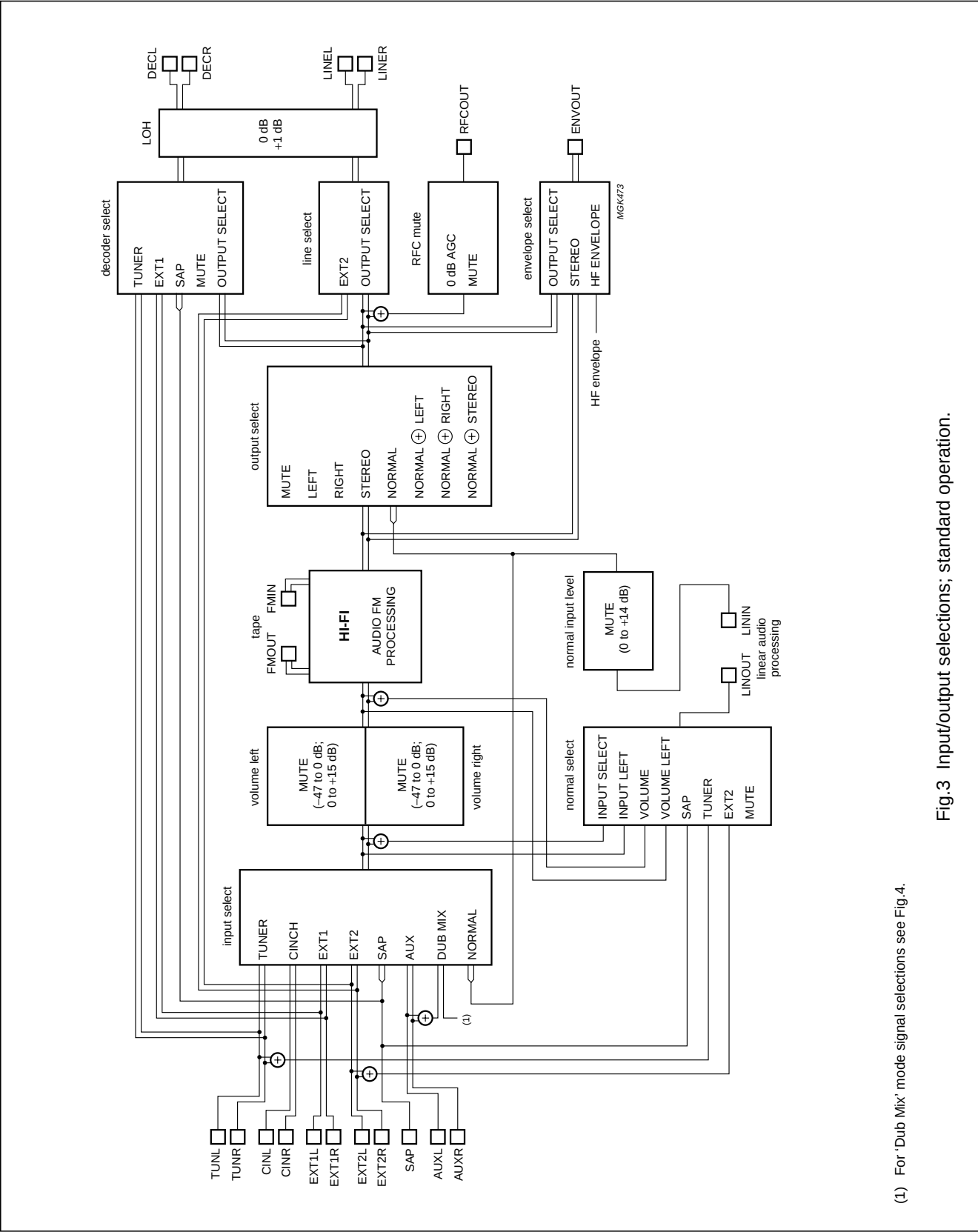
FUNCTIONAL DESCRIPTION

An overview of input/output selections is given in Figs 3 to 5.

Full control of the TDA9615H is accomplished via the 2-wire I²C-bus. Up to 400 kbits/s bus speed can be used, in accordance with the I²C-bus fast-mode specification. The detailed functional description can be found in Chapter "I²C-bus protocol".

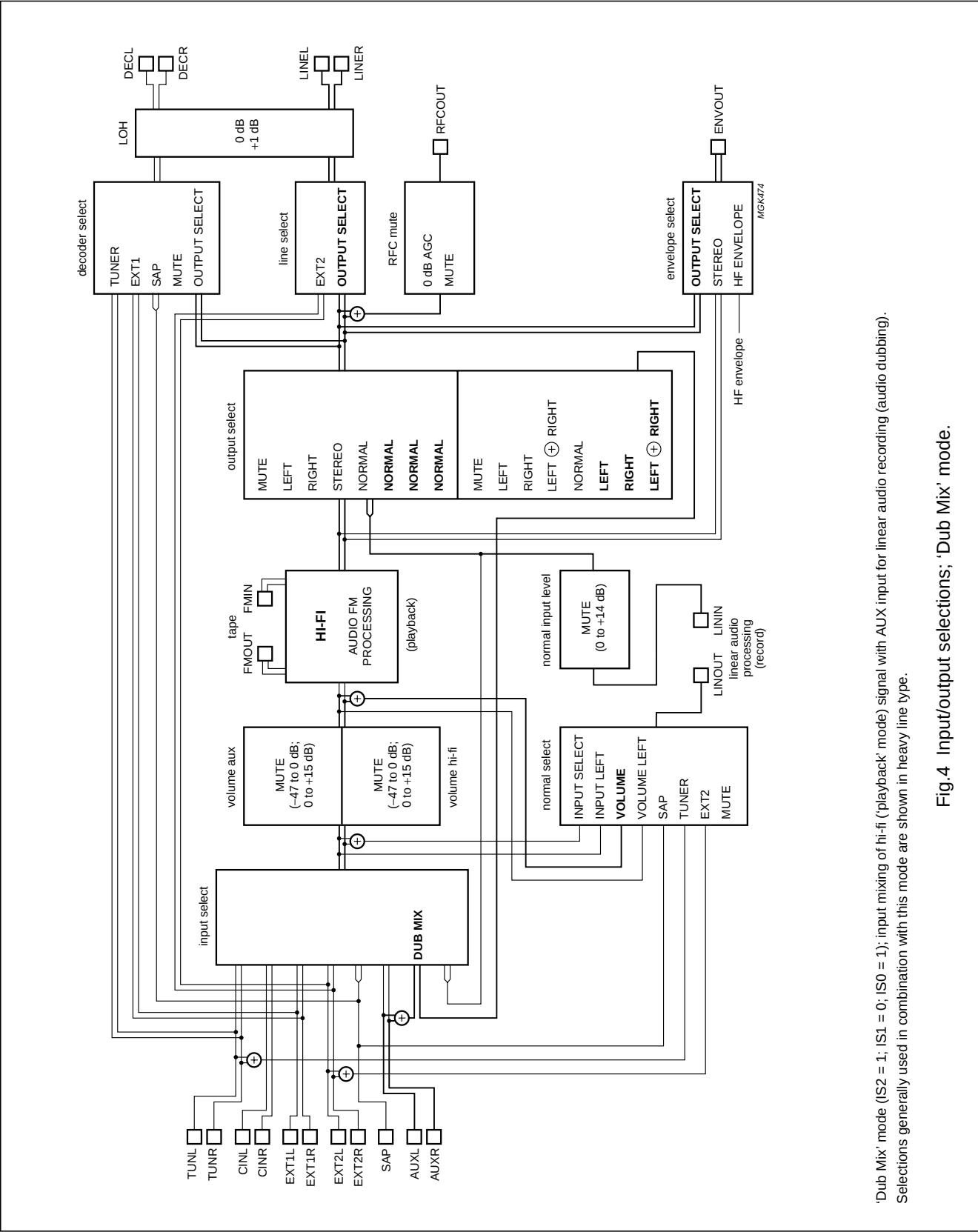
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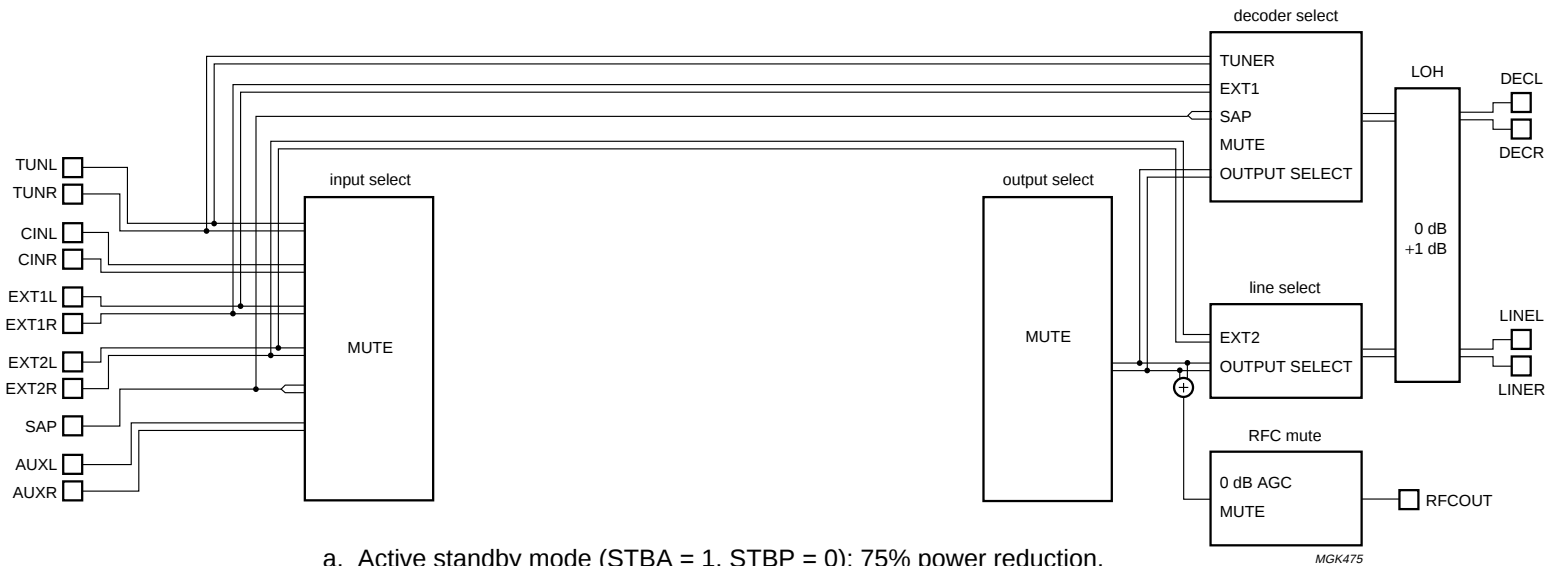


'Dub Mix' mode (IS2 = 1; IS0 = 0; ISO = 1); input mixing of hi-fi ('playback' mode) signal with AUX input for linear audio recording (audio dubbing).
Selections generally used in combination with this mode are shown in heavy line type.

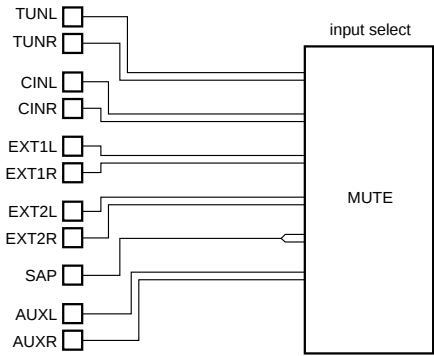
Fig.4 Input/output selections; 'Dub Mix' mode.

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a. Active standby mode (STBA = 1, STBP = 0); 75% power reduction.



b. Passive standby mode (STBP = 1); 90% power reduction.

Fig.5 Input/output selections; standby modes.

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I²C-BUS PROTOCOL**Addressing and data bytes**

For programming the device (write mode) seven data byte registers are available; they are addressable via eight subaddresses. Automatic subaddress incrementing enables the writing of successive data bytes in one transmission. During power-on, data byte registers are reset to a default state by use of a Power On Reset (POR) circuit which signal is derived from the internally generated I²C-bus supply voltage (V5OUT; pin 38). For reading from the device (read mode) one data byte register is available without subaddressing.

Table 1 TDA9615H addresses and data bytes

DATA BYTE	ADDRESS							
Write mode								
Slave address byte (B8H)	1	0	1	1	1	0	0	0
Subaddress bytes (00H to 07H)	0 ⁽¹⁾	0 ⁽¹⁾	0 ⁽¹⁾	0 ⁽¹⁾	0	0 or 1	0 or 1	0 or 1
Control byte (subaddress 00)	AFM	DOC	SHH	DETH	NTSC	MUTE	STBP	STBA
Select byte (subaddress 01)	DOS1	DOS0	s5	s4	NIL3	NIL2	NIL1	NIL0
Input byte (subaddress 02)	i7	IS2	IS1	IS0	NS2	NS1	NS0	i0
Output byte (subaddress 03)	LOH	OSN	OSR	OSL	EOS	LOS	DOS	RFCM
Left volume byte (subaddress 04)	l7	VLS	VL5	VL4	VL3	VL2	VL1	VL0
Right volume byte (subaddress 05)	r7	VRS	VR5	VR4	VR3	VR2	VR1	VR0
Volume byte (subaddress 06)	simultaneous loading of subaddress 04 and subaddress 05 register							
Power byte (subaddress 07)	CALS	VCCH	TEST	PORR	p3	p2	p1	p0
Read mode								
Slave address byte (B9H)	1	0	1	1	1	0	0	1
Read byte	CALR	AUTN	0 ⁽²⁾	POR	0 ⁽²⁾	1 ⁽²⁾	0 ⁽²⁾	0 ⁽²⁾

Notes

1. Use of subaddress F0H to F7H (1111 0XXX) instead of 00H to 07H (0000 0XXX) disables the automatic subaddress incrementing allowing continuous writing to a single data byte register.
2. The state of unused read bits are not reliable; their state may change during development.

Table 2 Status of data bytes after POR

DATA BYTE	ADDRESS							
Control byte	1	0	0	0	1	1	0	0
Select byte	0	0	1 ⁽¹⁾	1 ⁽¹⁾	0	0	0	0
Input byte	0 ⁽¹⁾	0	0	0	1	1	1	0 ⁽¹⁾
Output byte	0	0	0	0	0	0	0	1
Left volume byte	0 ⁽¹⁾	1	0	0	0	0	0	0
Right volume byte	0 ⁽¹⁾	1	0	0	0	0	0	0
Power byte	0	0	0	0	0 ⁽¹⁾	0 ⁽¹⁾	0 ⁽¹⁾	0 ⁽¹⁾

Note

1. For eventual future compatibility it is advised to keep unused write bits equal to POR state.

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Valid transmissions to and from TDA9615H

Table 3 Examples of valid transmissions

FUNCTION	DATA TRANSFER SEQUENCE
Write	START - B8H - 00H - data_for_00 - STOP
Write with auto-increment	START - B8H - 00H - data_for_00 - data_for_01 - data_for_02 - STOP
Auto-increment 'wrap-around'	START - B8H - 07H - data_for_07 - data_for_00 - data_for_01 - STOP
Write without auto-increment	START - B8H - F6H - data_for_06 - data_for_06 - data_for_06 - STOP
Read	START - B9H - data_from_ic - STOP
	START - B9H - data_from_ic - data_from_ic - data_from_ic - STOP

Overview of TDA9615H I²C-bus control**Table 4** Condensed overview

FUNCTION	MODES	CONTROL BITS
Audio FM mode	playback; loop-through ⁽¹⁾ ; record	AFM, DOC and SHH
Dropout cancelling	on; off	DOC
Headswitch noise cancel sample-and-hold time	6 µs; 8 µs	SHH
Playback hi-fi carrier detection	slow; fast	DETH
Record carrier ratio	0; 6; 8; 9.5; 11; 12.5; 13.5 dB	DOC, SHH and DETH
System standard	NTSC ⁽¹⁾ ; PAL	NTSC
Power mute	output muting ⁽¹⁾	MUTE
Operation mode	full operation ⁽¹⁾ ; active standby; passive standby	STBP and STBA
Normal input level	0 ⁽¹⁾ to +14 dB; mute	NIL3 to NIL0
Input select	Tuner ⁽¹⁾ ; Cinch; Ext1; Ext2; SAP; Dub Mix; Normal; Aux	IS2, IS1 and IS0
Normal select	Input Select; Volume; Input-Left; Volume-Left; SAP; Tuner; Ext2; mute ⁽¹⁾	NS2, NS1 and NS0
Line output amplification	0 dB ⁽¹⁾ ; +1 dB	LOH
Output select	mute ⁽¹⁾ ; Left; Right; Stereo; Normal; Mix-Left; Mix-Right; Mix-Stereo	OSN, OSR and OSL
Envelope output	Output Select ⁽¹⁾ ; Stereo; HF Envelope	EOS and AFM
Line output select	Output Select ⁽¹⁾ ; Ext2	LOS
Decoder output select	Output Select ⁽¹⁾ ; Tuner; Ext1; SAP; mute	DOS, DOS1 and DOS0
RFC output	0 dB; mute ⁽¹⁾	RFCM
Volume left	−47 to 0 dB ⁽¹⁾ ; 0 to +15 dB; mute	VLS, VL5 to VL0
Volume right	−47 to 0 dB ⁽¹⁾ ; 0 to +15 dB; mute	VRS, VR5 to VR0
Auto-calibration	off ⁽¹⁾ ; start calibration	CALS
Supply voltage	9 V ⁽¹⁾ ; 12 V	VCCH
Test	standard operation ⁽¹⁾ ; test modes	TEST, s4 to NIL0

Note

1. POR.

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Control byte; subaddress 00 (hi-fi and general control)**Table 5** Bits of control byte

BIT	DESCRIPTION
AFM	audio FM mode; see Table 6
DOC	dropout cancel; see Table 6
SHH	sample-and-hold high-state; see Table 6
DETH	detector high; see Table 6
NTSC	NTSC television system standard; see Table 7
MUTE	power mute; see Table 8
STBP	standby mode passive; see Table 9
STBA	standby mode active; see Table 9

Table 6 Bits AFM, DOC, SHH and DETH

AFM	DOC	SHH	DETH	MODE	REMARKS	DESCRIPTION
0	X ⁽¹⁾	X ⁽¹⁾	X ⁽¹⁾	playback ⁽²⁾		hi-fi circuit in playback mode
0	0	X ⁽¹⁾	X ⁽¹⁾	playback	DOC off	no dropout cancelling
0	1	X ⁽¹⁾	X ⁽¹⁾	playback	DOC on	dropout cancelling active
0	X ⁽¹⁾	0	X ⁽¹⁾	playback	sample-and-hold time = 6 μ s	headswitch noise cancel time is 6 μ s
0	X ⁽¹⁾	1	X ⁽¹⁾	playback	sample-and-hold time = 8 μ s	headswitch noise cancel time is 8 μ s
0	X ⁽¹⁾	X ⁽¹⁾	0	playback	detect = fast	hi-fi detector timing: fast mode
0	X ⁽¹⁾	X ⁽¹⁾	1	playback	detect = slow	hi-fi detector timing: slow mode
1	X ⁽¹⁾	X ⁽¹⁾	X ⁽¹⁾	record/loop-through		hi-fi circuit in record/loop-through mode
1	0	0	0	loop-through ⁽³⁾⁽⁴⁾		no FM output signal (EE mode)
1	0	0	1	record	0 dB mix	0 dB FM output carrier ratio (1 : 1)
1	0	1	0	record	6 dB mix	6 dB FM output carrier ratio (1 : 2)
1	0	1	1	record	8 dB mix	8 dB FM output carrier ratio (1 : 2.5)
1	1	0	0	record	9.5 dB mix	standard 9.5 dB FM output carrier ratio (1 : 3)
1	1	0	1	record	11 dB mix	11 dB FM output carrier ratio (1 : 3.5)
1	1	1	0	record	12.5 dB mix	12.5 dB FM output carrier ratio (1 : 4.2)
1	1	1	1	record	13.5 dB mix	13.5 dB FM output carrier ratio (1 : 4.7)

Notes

1. X = don't care.
2. Auto-normal function: if during hi-fi 'playback' mode no FM carrier is detected at FMIN (pin 37) the 'Normal' mode audio signal (LININ; pin 21) is automatically selected by the output select function.
 - a) Hi-fi carrier detection time (i.e. auto-normal release time) can be selected via bit DETH:
 'fast' mode: 1 to 2 HID periods (33 to 66 ms NTSC, 40 to 80 ms PAL)
 'slow' mode: 7 to 8 HID periods (233 to 267 ms NTSC, 280 to 320 ms PAL).
 - b) The status of hi-fi detection can be monitored via the I²C-bus; see bit AUTN of the read byte (see Table 31).
 - c) If muting is required instead of automatic selection of the 'Normal' mode audio signal the normal input level function should be set to mute; see bits NIL3 to NIL0 of the select byte.

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3. Modes 'loop-through' and 'record' are equal in audio signal flow; FMOUT (pin 36) however is muted during 'loop-through' mode.
4. POR.

Table 7 Bit NTSC

NTSC ⁽¹⁾	MODE	DESCRIPTION
0	PAL	hi-fi circuit in 'PAL' mode (FM carriers: 1.4 and 1.8 MHz)
1	NTSC ⁽²⁾	hi-fi circuit in 'NTSC' mode (FM carriers: 1.3 and 1.7 MHz)

Notes

1. Bit NTSC selects between the system standard settings for NTSC and PAL (SECAM) use. The auto-calibration function uses the system standard HID frequency of 29.97 Hz for NTSC and 25 Hz for PAL. After calibration bit NTSC allows immediate switching between the NTSC and PAL system standard.
2. POR.

Table 8 Bit MUTE

MUTE ⁽¹⁾	MODE	DESCRIPTION
0	–	power mute function released; mute switches open
1	mute ⁽²⁾	power mute function activated; mute switches closed

Notes

1. Bit MUTE controls the line and RFC output mute switches at pins 13, 14 and 17 (power mute function). Power mute is also automatically activated at supply voltage power-up or power-down (V_{CC} ; pin 35).
2. POR.

Table 9 Bits STBP and STBA

STBP	STBA	MODE	DESCRIPTION
0	0	– (note 1)	full operation
0	1 ⁽²⁾	active standby ⁽³⁾	standby mode; reduced power consumption
1 ⁽⁴⁾	X ⁽⁵⁾	passive standby ⁽³⁾	standby mode; minimum power consumption

Notes

1. POR.
2. By selecting STBA = 1 the TDA9615H is switched to low-power 'active standby' mode. To reduce power consumption most circuits are switched off. RFC, line and decoder outputs however remain active. This way the direct audio selections offered via the line output select and decoder output select functions (bits LOS and DOS of the output byte) remain operable in this mode. The 'Output Select' mode signal is muted.
3. Calibration and I²C-bus registers are not affected by using 'active standby' or 'passive standby' mode.
4. By selecting STBP = 1 the TDA9615H is switched to minimum power 'passive standby' mode. All circuits except power mute, I²C-bus and the line input reference buffer (voltage at pins 1 to 10 and 44) are switched off for minimum power consumption. Use of the power mute function (bit MUTE of control byte) ensures pop-free switching of the line and RFC output to and from 'passive standby' mode. To obtain minimum power consumption the power mute function should be de-activated again during 'passive standby' mode.
5. X = don't care.

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Select byte; subaddress 01 (decoder output select and linear audio volume control)**Table 10** Bits of select byte

BIT	DESCRIPTION
DOS1	decoder output select 1; see Table 11
DOS0	decoder output select 0; see Table 11
NIL3	normal input level 3; see Table 12
NIL2	normal input level 2; see Table 12
NIL1	normal input level 1; see Table 12
NIL0	normal input level 0; see Table 12

Table 11 Bits DOS1 and DOS0; note 1

DOS1	DOS0	MODE	DESCRIPTION
0	0	Tuner	decoder output signal is TUNL and TUNR input signal
0	1	Ext1	decoder output signal is EXT1L and EXT1R input signal
1	0	SAP	decoder output signal is SAP input signal
1	1	mute	mute

Note

1. By selecting bit DOS = 1 of the output byte several independent signal input selections are offered for the decoder outputs DECL and DECR (pins 18 and 19) via bits DOS1 and DOS0:
 - a) TUNL and TUNR inputs (pins 1 and 2)
 - b) EXT1L and EXT1R inputs (pins 5 and 6)
 - c) SAP input (pin 44)
 - d) Mute.

These decoder selections are also operable in 'active standby' mode (bit STBA = 1 of the control byte).

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Table 12 Bits NIL3, NIL2, NIL1 and NIL0; note 1

NIL3	NIL2	NIL1	NIL0	DESCRIPTION
0	0	0	0	0 dB; note 2
0	0	0	1	1 dB
0	0	1	0	2 dB
0	0	1	1	3 dB
0	1	0	0	4 dB
0	1	0	1	5 dB
0	1	1	0	6 dB
0	1	1	1	7 dB
1	0	0	0	8 dB
1	0	0	1	9 dB
1	0	1	0	10 dB
1	0	1	1	11 dB
1	1	0	0	12 dB
1	1	0	1	13 dB
1	1	1	0	14 dB
1	1	1	1	mute

Notes

1. Mute and 15 settings of amplification can be selected for the linear audio input signal (LININ; pin 21). This level control can replace the manual adjustment of 'playback' mode level at the linear audio circuit.
2. POR.

Input byte; subaddress 02 (input selection for hi-fi and normal audio)**Table 13** Bits of input byte

BIT	DESCRIPTION
IS2	input select 2; see Table 14
IS1	input select 1; see Table 14
IS0	input select 0; see Table 14
NS2	normal select 2; see Table 15
NS1	normal select 1; see Table 15
NS0	normal select 0; see Table 15

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Table 14 Bits IS2, IS1 and IS0; note 1

IS2	IS1	IS0	MODE	DESCRIPTION
0	0	0	Tuner ⁽²⁾	TUNL and TUNR input
0	0	1	Cinch	CINL and CINR input
0	1	0	Ext1	EXT1L and EXT1R input (TV input)
0	1	1	Ext2	EXT2L and EXT2R input (decoder input)
1	0	0	SAP	SAP; mono input
1	0	1	Dub Mix ⁽³⁾	AUX input signal (L) and selection of hi-fi output signal (R)
1	1	0	Normal	LININ input (linear audio)
1	1	1	Aux	AUXL and AUXR input (e.g. camcorder input)

Notes

- Bits IS2, IS1 and IS0 select the input signal which is led to the volume controls of the hi-fi processing and generally via the normal select function to the linear audio circuit (LINOUT; pin 20).
- POR.
- 'Dub Mix' is a special selection for linear audio use supporting audio dubbing (a video recorder feature mode for sound recording of linear audio only). 'Dub Mix' connects the AUX input signal to the left channel ($\frac{1}{2}$ AUXL + $\frac{1}{2}$ AUXR) and the hi-fi output signal to the right channel (generally $\frac{1}{2}$ hi-fi left + $\frac{1}{2}$ hi-fi right but also hi-fi left or hi-fi right can be selected). 'Dub Mix' also changes part of the 'Output Select' mode settings to 'Normal' mode for monitoring of the linear audio recording (see control bits OSN, OSR and OSL of the output byte for 'Dub Mix' mode input and output selection).

Table 15 Bits NS2, NS1, NS0; note 1

NS2	NS1	NS0	MODE	DESCRIPTION
0	0	0	Input Select	hi-fi input selection
0	0	1	Volume	hi-fi volume control
0	1	0	Input-Left	left channel of hi-fi input selection only
0	1	1	Volume-Left	left channel of hi-fi volume control
1	0	0	SAP	SAP; mono input
1	0	1	Tuner	TUNL and TUNR input
1	1	0	Ext2	EXT2L and EXT2R input
1	1	1	mute ⁽²⁾	mute

Notes

- With bits NS2, NS1 and NS0 the input signal is selected which is available at the LINOUT output (pin 20) for connection to an external linear audio circuit. The signal selected with the input select function (via bits IS2, IS1 and IS0) is available in the following ways:
 - Left and right channel combined, optional including hi-fi volume control ($\frac{1}{2}$ hi-fi left + $\frac{1}{2}$ hi-fi right).
 - Left channel only (language I), optional including left channel volume control.
 Furthermore the SAP input (pin 44), tuner input (pins 1 and 2) or EXT2 input (pins 7 and 8) can be selected independently. Also a mute setting is available.
- POR.

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Output byte; subaddress 03 (output selection and control)**Table 16** Bits of output byte

BIT	DESCRIPTION
LOH	line output high; see Table 17
OSN	output select normal; see Table 18
OSR	output select right; see Table 18
OSL	output select left; see Table 18
EOS	envelope output select; see Table 20
LOS	line output select; see Table 21
DOS	decoder output select; see Table 22
RFCM	RFC mute; see Table 23

Table 17 Bit LOH; note 1

LOH	MODE	DESCRIPTION
0	0 dB ⁽²⁾	no line output amplification
1	1 dB	1 dB line output amplification

Notes

1. An additional 1 dB amplification can be selected for line and decoder outputs LINEL, LINER, DECL and DECR (pins 15, 16, 18 and 19).
2. POR.

Table 18 Bits OSN, OSR and OSL; notes 1 and 2

OSN	OSR	OSL	MODE	DESCRIPTION
0	0	0	mute ⁽³⁾	mute; no selection
0	0	1	Left	left hi-fi channel selected (language I)
0	1	0	Right	right hi-fi channel selected (language II)
0	1	1	Stereo	hi-fi stereo selected
1	0	0	Normal	normal selected (linear audio; LININ input)
1	0	1	Mix-Left	mix of hi-fi left with normal ($\frac{1}{2}$ hi-fi left + $\frac{1}{2}$ normal)
1	1	0	Mix-Right	mix of hi-fi right with normal ($\frac{1}{2}$ hi-fi right + $\frac{1}{2}$ normal)
1	1	1	Mix-Stereo	mix of hi-fi stereo with normal ($\frac{1}{2}$ hi-fi stereo + $\frac{1}{2}$ normal)

Notes

1. When no hi-fi signal is found on tape during 'playback' mode the auto-normal function is activated; all output select function modes except 'mute' will be overruled and changed to 'Normal' mode. If muting of the hi-fi sound is desired instead of selecting linear audio the normal input level (bits NIL3 to NIL0 of the select byte) should be set to mute. Activation of the auto-normal function can be monitored by reading bit AUTN of the read byte.
2. If 'Dub Mix' mode is selected via the input select function (see control bits IS2, IS1 and IS0 of the input byte) functionality of the modes 'Mix-Left', 'Mix-Right' and 'Mix-Stereo' are changed to support audio dubbing input mixing. Hi-fi channel selection is offered for the input and normal sound is available at the output for monitoring of the linear audio recording. See also Table 19.
3. POR.

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Table 19 Dub Mix selections

MODE	DUB MIX OUTPUT SELECTION	DUB MIX INPUT SELECTION
mute ⁽¹⁾	mute	mute
Left	left	hi-fi left
Right	right	hi-fi right
Stereo	stereo	$\frac{1}{2}$ hi-fi left + $\frac{1}{2}$ hi-fi right
Normal	normal	normal
Mix-Left	normal	hi-fi left
Mix-Right	normal	hi-fi right
Mix-Stereo	normal	$\frac{1}{2}$ hi-fi left + $\frac{1}{2}$ hi-fi right

Note

1. POR.

Table 20 Bit EOS

EOS ⁽¹⁾	MODE	DESCRIPTION
0	Output Select ⁽²⁾	audio peak envelope of 'Output Select' mode signal
1	Stereo or HF envelope ⁽³⁾	audio peak envelope of hi-fi stereo or HF envelope

Notes

1. A signal selection for output ENVOUT (pin 39) is available using bit EOS.
2. POR.
3. The selection made for mode 'Stereo or HF envelope' depends upon the mode of the hi-fi processing:
 - a) Hi-fi 'loop-through' mode or 'record' mode (bit AFM = 1): audio peak envelope of hi-fi stereo.
 - b) Hi-fi 'playback' mode (bit AFM = 0): HF envelope of left channel FM carrier.

Table 21 Bit LOS; notes 1 and 2

LOS	MODE	DESCRIPTION
0	Output Select ⁽³⁾	line output signal is set by output select function
1	Ext2	line output signal is EXT2L and EXT2R input

Notes

1. An independent selection of EXT2L and EXT2R inputs (pins 7 and 8) to the line outputs LINEL and LINER (pins 15 and 16) is possible by setting bit LOS to 1. This direct selection is also operable in 'active standby' mode (STBA = 1 of the control byte).
2. Using bit DOS and bits DOS1 and DOS0 of the output byte mode 'Tuner' or 'Ext1' selection to the outputs DECL and DECR (pins 18 and 19) is offered realizing full Euro-SCART pay-TV decoder switching.
3. POR.

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Table 22 Bit DOS; notes 1 and 2

DOS	MODE	DESCRIPTION
0	Output Select ⁽³⁾	decoder output signal is set by output select function
1	decoder select ⁽⁴⁾	decoder output signal is set by decoder output select function

Notes

1. An independent selection of TUNL and TUNR (pins 1 and 2), EXT1L and EXT1R (pins 5 and 6), SAP (pin 44) or mute is possible for the decoder outputs DECL and DECR (pins 18 and 19) by setting bit DOS to 1. Source selection is offered by using bits DOS1 and DOS0 of the select byte. These direct selections are also operable in 'active standby' mode (STBA = 1 of the control byte).
2. In combination with the independent EXT2 selection to the outputs LINEL and LINER (pins 15 and 16) via bit LOS full Euro-SCART pay-TV decoder switching is offered.
3. POR.
4. Internal mode.

Table 23 Bit RFCM; note 1

RFCM	MODE	DESCRIPTION
0	–	RFC output is set by output select function
1	mute ⁽²⁾	RFC output signal is muted (AGC reset)

Notes

1. RF converter output RFCOUT (pin 12) can be muted using bit RFCM. During mute the AGC capacitor at pin 11 (RFCAGC) is discharged, resetting AGC control.
2. POR.

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Volume bytes; subaddresses 04, 05 and 06 (left and right channel volume control)

Table 24 Bits of volume bytes; see Table 25

BIT	DESCRIPTION
VLS	volume left sign
VL5	volume left 5
↓	↓
VL0	volume left 0
VRS	volume right sign
VR5	volume right 5
↓	↓
VR0	volume right 0

Table 25 bits VLS, VRS, VL5 to VL0 and VR5 to VR0; note 1

VLS	VL5	VL4	VL3	VL2	VL1	VL0	MODE
VRS	VR5	VR4	VR3	VR2	VR1	VR0	
0	0	0	0	0	0	0	0 dB
0	0	0	0	0	0	1	−1 dB
0	0	0	0	0	1	0	−2 dB
0	0	0	0	0	1	1	−3 dB
0	0	0	0	1	0	0	−4 dB
↓	↓	↓	↓	↓	↓	↓	↓
0	1	0	1	0	1	1	−43 dB
0	1	0	1	1	0	0	−44 dB
0	1	0	1	1	0	1	−45 dB
0	1	0	1	1	1	0	−46 dB
0	1	0	1	1	1	1	−47 dB
0	1	1	X ⁽²⁾	X ⁽²⁾	X ⁽²⁾	X ⁽²⁾	mute
1	X ⁽²⁾	X ⁽²⁾	0	0	0	0	0 dB ⁽³⁾
1	X ⁽²⁾	X ⁽²⁾	0	0	0	1	+1 dB
1	X ⁽²⁾	X ⁽²⁾	0	0	1	0	+2 dB
1	X ⁽²⁾	X ⁽²⁾	0	0	1	1	+3 dB
1	X ⁽²⁾	X ⁽²⁾	0	1	0	0	+4 dB
↓	↓	↓	↓	↓	↓	↓	↓
1	X ⁽²⁾	X ⁽²⁾	1	0	1	1	+11 dB
1	X ⁽²⁾	X ⁽²⁾	1	1	0	0	+12 dB
1	X ⁽²⁾	X ⁽²⁾	1	1	0	1	+13 dB
1	X ⁽²⁾	X ⁽²⁾	1	1	1	0	+14 dB
1	X ⁽²⁾	X ⁽²⁾	1	1	1	1	+15 dB

Notes

- Left and right volume controls can be set simultaneous by using subaddress 06 (volume byte). Addressing the volume byte will load both the left and right volume registers with the same data value.
- X = don't care.
- POR.

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Power byte; subaddress 07 (power-up control and test)**Table 26** Bits of power byte

BIT	DESCRIPTION
CALS	calibration start; see Table 27
VCCH	V _{CC} high; see Table 28
TEST	test; see Table 29
PORR	POR reset; see Table 30

Table 27 Bit CALS; note 1

CALS⁽²⁾	MODE	DESCRIPTION
0	– (note 3)	–
1	start calibration	start of automatic calibration cycle

Notes

1. Pin 39 (ENVOUT) or bit CALR of the read byte can be monitored to check for successful completion of the calibration. It is advised to keep CALS logic 1 after calibration to ensure a reliable CALR state.
2. After a change of bit CALS from logic 0 to logic 1, an automatic frequency calibration is performed setting hi-fi modem, band-pass filter and noise reduction. Use of auto-calibration is only needed after power-up of the supply voltage (POR).
3. POR.

Table 28 Bit VCCH; notes 1 and 2

VCCH	MODE	DESCRIPTION
0	9 V; note 3	output DC voltage is 4.5 V
1	12 V	output DC voltage is 6 V

Notes

1. To maximize the output drive when using 12 V supply voltage (V_{CC}; pin 35), the DC level of outputs LINEL, LINER, DECL and DECR (pins 15, 16, 18 and 19) can be changed using bit VCCH.
2. Use of the power mute function (bit MUTE of the control byte) ensures disturbance-free switching of the line output signal when setting VCCH after power-up.
3. POR.

Table 29 Bit TEST; note 1

TEST	MODE	DESCRIPTION
0	– (note 2)	standard operation
1	test	special measurement test modes

Notes

1. Several special test modes can be selected for testing and evaluation purposes. Bits s4 to NIL0 of the select byte are used for selection of these tests.
2. POR.

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Table 30 Bit PORR; note 1

PORR	MODE	DESCRIPTION
0	– (note 2)	–
1	POR reset	reset of POR signal bit

Notes

1. Bit POR of the read byte can be used to detect the occurrence of a power-on reset situation (bit POR reads logic 1 in case the I²C register contents equal the POR default state). When applying bit POR this way, PORR should be used after a power-on reset to force a reset of the POR read bit state to logic 0.
2. POR.

Read byte (device state information)**Table 31** Bits of read byte

BIT	DESCRIPTION
CALR	calibration ready; see Table 32
AUTN	auto-normal; see Table 33
POR	power-on reset; see Table 34

Table 32 Bit CALR; notes 1 and 2

CALR	DESCRIPTION
0	device is not calibrated
1	device has completed auto-calibration

Notes

1. Bit CALR = 0 indicates the device is not calibrated. After an automatic calibration is started by setting bit CALS of the power byte the end of calibration is signalled by this bit. If for some reason a calibration cannot be completed (e.g. no HID signal available or hi-fi processing is in 'playback' mode) CALR remains logic '0'. If calibration is lost due to a power-on reset situation CALR will return to logic 0 as well.
2. Also the envelope output (pin 39) can be used to check for end of calibration.

Table 33 Bit AUTN; notes 1 and 2

AUTN	DESCRIPTION
0	audio FM signal is detected at FM input
1	no audio FM signal detected; normal sound is selected instead of hi-fi

Notes

1. When hi-fi processing is in 'playback' mode but no hi-fi carrier input signal is detected the auto-normal function is activated. Auto-normal overrules the 'Output Select' mode setting, selecting normal sound (i.e. linear audio) instead of hi-fi. The state of this auto-normal function can be checked via bit AUTN.
2. The auto-normal function is only active (bit AUTN = logic 1) for hi-fi 'playback' mode (bit AFM = 0); during 'record' or 'loop-through' mode bit AUTN is always logic 0.

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Table 34 Bit POR; notes 1 and 2

POR	DESCRIPTION
0	I ² C-bus bit state differs from power-on reset state
1	I ² C-bus bit state equals power-on reset state

Notes

1. At power-on or during a power voltage dip, an internal power-on reset signal is generated which resets the I²C-bus data bits to a predefined state. When the internal data bits are found to be in POR state (due to an actual power-on reset but also if set via the I²C-bus) bit POR signals logic 1.
2. Using the POR bit to detect the occurrence of a power-on reset requires bit PORR to be set logic 1 after power-up. Setting bit PORR forces the POR bit to logic 0 independent of other I²C-bus bit settings. After calibration is completed also the CALR bit can be used to detect a power-on reset; calibration will be lost in such situation which is signalled by CALR = 0.

LIMITING VALUES

In accordance with the Absolute Maximum Rating System (IEC 134).

SYMBOL	PARAMETER	MIN.	MAX.	UNIT
V _{CC}	supply voltage	0	13.2	V
T _{stg}	storage temperature	−65	+150	°C
T _{amb}	operating ambient temperature	0	70	°C
V _{es}	electrostatic handling			
	machine model	−300	+300	V
	human body model	−3000	+3000	V

THERMAL CHARACTERISTICS

SYMBOL	PARAMETER	CONDITIONS	VALUE	UNIT
R _{th j-a}	thermal resistance from junction to ambient	in free air	60	K/W

DC CHARACTERISTICS

SYMBOL	PARAMETER	CONDITIONS	MIN.	TYP.	MAX.	UNIT
Supply voltage						
V _{CC}	supply voltage; pin 35	VCCH = 0	8.1	9	13.2	V
		VCCH = 1	8.1	12	13.2	V
Supply current						
I _{CC}	supply current; pin 35		–	45	60	mA
		STBA = 1 (‘active standby’ mode)	–	12	18	mA
		STBP = 1 (‘passive standby’ mode)	–	4	6	mA

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SYMBOL	PARAMETER	CONDITIONS	MIN.	TYP.	MAX.	UNIT
Input voltages (internally generated)						
V_I	input voltage		–	3.8	–	V
	pins 1 to 10, 21 and 44 pin 37		–	1.9	–	V
Output voltages						
V_O	output voltage		–	3.8	–	V
	pin 12		–	4.5	–	V
	pins 15, 16, 18 and 19	VCCH = 0	–	6	–	V
	pin 20	VCCH = 1	–	4.5	–	V
	pin 36		–	1.2	–	V
	pin 38		–	5	–	V
Output current						
I_O	output current at pin 38		–	–	3	mA
HID input; pin 40						
V_{IH}	HIGH-level input voltage		2.75	–	5.5	V
V_{IL}	LOW-level input voltage		0	–	2.25	V

AC CHARACTERISTICS**Loop-through mode**

Audio input level –8 dBV for $f = 1$ kHz at TUN inputs (pins 1 and 2); $V_{CC} = 12$ V; $T_{amb} = 25$ °C; POR state with mute off; calibrated; bit NTSC set to logic 1 or logic 0; measured in application diagram of Fig.6; unless otherwise specified.

SYMBOL	PARAMETER	CONDITIONS	MIN.	TYP.	MAX.	UNIT
Line inputs and linear audio input (TUN, CIN, EXT1, EXT2, AUX, SAP and LININ; pins 1 to 10, 44 and 21)						
R_i	input impedance		100	130	–	k Ω
V_i	input voltage		–	–	9	dBV
Line and decoder outputs (LINEL, LINER, DECL, DECR; pins 15, 16, 18 and 19)						
V_o	output voltage		–9	–8	–7	dBV
		LOH = 1 (+1 dB output signal)	–8	–7	–6	dBV
$V_{o(max)}$	maximum output voltage	note 1	7	8	–	dBV
		note 1; VCCH = 1; $V_{CC} = 12$ V	10	11	–	dBV
THD	total harmonic distortion		–	0.01	0.1	%
α_{cb}	channel balance		–1	0	+1	dB
V_n	noise level	TUN = $-\infty$ dBV; note 2	–	–94	–90	dBV
R_o	output impedance		–	100	150	Ω
α_{mute}	volume mute	VLS to VL0 and VRS to VR0 = mute	–	–95	–70	dB
	output mute	OSN, OSR and OSL = mute	–	–85	–70	dB

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SYMBOL	PARAMETER	CONDITIONS	MIN.	TYP.	MAX.	UNIT
$\alpha_{ct(ch)}$	crosstalk between channels	channel crosstalk TUNL or TUNR = $-\infty$ dBV	–	–83	–70	dB
$\alpha_{ct(ch)(i)}$	crosstalk between input channels	note 3	–	–93	–	dB
$I_{o(max)}$	output current limiting	MUTE = 1; power mute	–	+1	–	mA
I_o	discharge output current	STBP = 1; passive standby	–	–1	–	mA
Linear audio output (LINOUT; pin 20)						
V_o	output voltage		–9	–8	–7	dBV
R_o	output impedance		–	200	300	Ω
RF converter output (RFCOUT; pin 12)						
V_o	output voltage		–9	–8	–7	dBV
	output voltage at high level	TUN = +8 dBV	–4.5	–3	–1.5	dBV
THD	total harmonic distortion		–	0.03	–	%
		TUN = up to +8 dBV	–	<0.2	–	%
V_n	noise level	TUN = $-\infty$ dBV; note 2	–	–80	–	dBV
R_o	output impedance		–	200	300	Ω
$I_{o(max)}$	output current limiting	MUTE = 1; power mute	–	+1	–	mA
I_o	discharge output current	STBP = 1; passive standby	–	–1	–	mA
Power mute outputs (MUTEC, MUTEL, MUTER; pins 13, 14 and 17)						
R_o	output impedance		50	100	–	k Ω
		MUTE = 1; DC load = ± 1 mA	–	15	–	Ω
Envelope output (ENVOUT; pin 39); audio peak envelope						
V_o	output voltage		1.69	1.8	1.91	V
	output voltage at zero level	TUN = $-\infty$ dBV	–	–	0.3	V
	output voltage at zero level; maximum volume	TUN = $-\infty$ dBV; volume = +15 dB	–	–	0.35	V
	output voltage at high level	TUN = +8 dBV	4.0	4.5	5.0	V
α_{cb}	channel balance		–0.11	0	+0.11	V
R_o	output impedance		–	1	1.5	k Ω

Notes

1. THD = 1%; output load: $R_L = 5$ k Ω ; $C_L = 2.2$ nF; volume = +3 dB (12 V); TUN level varied.
2. Typical value: B = 20 Hz to 20 kHz, 'unweighted'; production testing: B = 300 Hz to 20 kHz, 'unweighted'.
3. Crosstalk of any line input pair (TUN, CIN, EXT1, EXT2, AUX or SAP) to any other line input.

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Record mode

For test modes see Table 38.

SYMBOL	PARAMETER	CONDITIONS	MIN.	TYP.	MAX.	UNIT
Noise reduction (test 25: LINEL and LINER; pins 15 and 16)						
THD	total harmonic distortion		–	0.1	0.3	%
		at high level; TUN = +8 dBV	–	0.2	1	%
α_{cb}	channel balance		–1	0	+1	dB
α_{lin}	linearity	TUN = –8 to –68 dBV	–31.5	–30.2	–28.5	dB
	high-level linearity	TUN = –8 to +8 dBV	7.5	8	8.5	dB
α_n	noise level with respect to output signal	TUN = –8 to $-\infty$ dBV; note 1	–	–46	–41	dB
t_{att}	attack time	in accordance with VHS specification	–	5	–	ms
t_{rec}	recovery time		–	70	–	ms
f_{res}	frequency response 300 Hz	TUN = 1 kHz to 300 Hz	–0.7	–0.2	+0.3	dB
	frequency response 10 kHz	TUN = 1 kHz to 10 kHz	3.1	3.9	4.7	dB
	audio low-pass filter response 20 kHz	TUN = 1 kHz to 20 kHz; test 26	–0.5	–0.1	+0.5	dB
	audio low-pass filter response 60 kHz	TUN = 1 kHz to 60 kHz; test 26	–	–24	–12	dB
FM modulator (test 25, test 26 and no test: FMOUT; pin 36)						
THD	total harmonic distortion	$\Delta f = 50$ kHz	–	0.1	0.2	%
$\Delta f_{(max)}$	maximum FM frequency deviation		140	150	160	kHz
$f_{c(error)}$	carrier frequency error	after calibration	–5	0	+5	kHz
$\Delta f_{c(l-r)}$	carrier frequency difference between left and right channel	$f_{c(r)} - f_{c(l)}$ after NTSC calibration	399.2	401.2	403.2	kHz
Δf_c	carrier frequency shift	NTSC/PAL system switching	97	100	103	kHz
TC	temperature coefficient		–	± 50	–	ppm/K
Noise reduction and FM modulator (FMOUT; pin 36)						
Δf	FM deviation		44.5	50	56.1	kHz
HF output (FMOUT; pin 36)						
$V_{o(rms)}$	output voltage left carrier	1st harmonic (RMS value)	53	60	68	mV
	output voltage right carrier	1st harmonic (RMS value)	160	180	202	mV
$V_{o(l)}/V_{o(r)}$	output voltage ratio between left and right carrier		2.7	3.0	3.3	
D_{3H}	distortion (3rd harmonic)	3rd harmonic/1st harmonic	–	–28	–18	dB
R_o	output impedance		–	100	150	Ω

Note

1. Typical value: B = 20 Hz to 20 kHz, 'unweighted'; production testing: B = 300 Hz to 20 kHz, 'unweighted'.

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Playback mode

$V_{\text{FMIN(rms)}} = 30 \text{ mV}$ (RMS value); $\Delta f = 50 \text{ kHz}$; $f_{\text{mod}} = 1 \text{ kHz}$; $f = f_0$; $V_{\text{CC}} = 12 \text{ V}$; $T_{\text{amb}} = 25 \text{ }^\circ\text{C}$; bit NTSC set to logic 1 or logic 0; measured in application diagram of Fig.6; unless otherwise specified. For test modes see Table 38.

SYMBOL	PARAMETER	CONDITIONS	MIN.	TYP.	MAX.	UNIT
FMIN; pin 37 and HF AGC (test 5 and test 6: FMOUT; pin 36)						
$V_{i(p-p)}$	input voltage level (peak-to-peak value)	left plus right channel; note 1	–	0.17	2	V
	AGC start level (at pin 37)	left plus right channel	59	84	118	mV
B	control bandwidth	note 2	–	10	–	kHz
Left channel band-pass filter (test 3: FMOUT; pin 36)						
V_o	band-pass filter output voltage	$f = f_0$; $V_{\text{FMIN(rms)}} = 30 \text{ mV}$ (RMS value)	105	150	215	mV
V_{o1}/V_o	band-pass filter output voltage ratio	$\frac{f_0 - 400 \text{ kHz}}{f_0}$	–	–30	–20	dB
V_{o2}/V_o		$\frac{f_0 - 150 \text{ kHz}}{f_0}$	–9	–6	–	dB
V_{o3}/V_o		$\frac{f_0 + 150 \text{ kHz}}{f_0}$	–9	–5	–	dB
V_{o4}/V_o		$\frac{f_0 + 250 \text{ kHz}}{f_0}$	–	–17	–12	dB
V_{o5}/V_o		$\frac{f_0 + 250 \text{ kHz}}{f_0 + 150 \text{ kHz}}$	–	–12	–9	dB
V_{o6}/V_o		$\frac{f_0 + 400 \text{ kHz}}{f_0}$	–	–	–30	dB
$R_{d(g)}$	group delay ripple	$f_0 - 150 \text{ kHz}$ to $f_0 + 150 \text{ kHz}$	–	<0.5	–	μs

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SYMBOL	PARAMETER	CONDITIONS	MIN.	TYP.	MAX.	UNIT
Right channel band-pass filter (test 4: FMOUT; pin 36)						
V_o	band-pass filter output voltage	$f = f_0$; $V_{FMIN(rms)} = 30$ mV (RMS value)	105	150	215	mV
V_{o1}/V_o	band-pass filter output voltage ratio	$\frac{f_0 - 400 \text{ kHz}}{f_0}$	–	–	–30	dB
V_{o2}/V_o		$\frac{f_0 - 250 \text{ kHz}}{f_0}$	–	–17	–10	dB
V_{o3}/V_o		$\frac{f_0 - 150 \text{ kHz}}{f_0}$	–9	–5	–	dB
V_{o4}/V_o		$\frac{f_0 - 250 \text{ kHz}}{f_0 - 150 \text{ kHz}}$	–	–12	–9	dB
V_{o5}/V_o		$\frac{f_0 + 150 \text{ kHz}}{f_0}$	–9	–6	–	dB
V_{o6}/V_o		$\frac{f_0 + 400 \text{ kHz}}{f_0}$	–	–30	–20	dB
$R_{d(g)}$	group delay ripple	$f_0 - 150 \text{ kHz}$ to $f_0 + 150 \text{ kHz}$	–	<0.5	–	μ s
Hi-fi detector and dropout cancelling						
$V_{i(A)}$	AUTN activation level	left channel carrier at FMIN (RMS value)	3.0	4.2	5.8	mV
$V_{i(D)}/V_{i(A)}$	DOC level with respect to AUTN level	left channel carrier at FMIN	–7	–4	–2	dB
$t_{d(A)}$	AUTN activation delay	carrier to no carrier at FMIN	200	300	500	μ s
$t_{d(D)}$	DOC switch-off delay	no carrier to carrier at FMIN	5	9	14	μ s
Head switching noise suppressor (test 25: LINEL and LINER; pins 15 and 16)						
t_h	hold pulse length		5	6	7	μ s
		SHH = 1	7	8	9	μ s
THD	total harmonic distortion (audio signal)	note 3	–	–	–73	dB
t_d	delay from HID to hold		–	0.3	–	μ s
Envelope output (ENVOUT; pin 39)						
V_o	output voltage	FMIN = 2 mV (RMS value); left channel	0.6	0.9	1.2	V
		FMIN = 20 mV (RMS value); left channel	2.5	2.9	3.3	V
		FMIN = 200 mV (RMS value); left channel	4.2	4.7	5.0	V

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SYMBOL	PARAMETER	CONDITIONS	MIN.	TYP.	MAX.	UNIT
PLL FM demodulator (test 25: LINEL and LINER; pins 15 and 16)						
V_i	sensitivity	$\Delta f = 150 \text{ kHz}$; $S/N = 35 \text{ dB}$	–	0.3	1.2	mV
THD	total harmonic distortion	$\Delta f = 50 \text{ kHz}$	–	0.03	0.3	%
	total harmonic distortion at maximum	$\Delta f = 150 \text{ kHz}$	–	0.2	1.5	%
S/N	signal to noise ratio	$\Delta f = 50 \text{ to } 0 \text{ kHz}$	54	60	–	dB
$\alpha_{ct(ch)}$	channel crosstalk	left or right carrier $\Delta f = 0 \text{ kHz}$	–	–80	–	dB
Noise reduction (test 26: LINEL and LINER; pins 15 and 16)						
V_n	noise level	TUN = $-\infty \text{ dBV}$; note 4	–	–96	–90	dBV
THD	total harmonic distortion	TUN = -3.5 dBV	–	0.05	0.2	%
α_l	linearity	TUN = $-3.5 \text{ to } -33.5 \text{ dBV}$	58	59.6	62	dB
f_{res}	frequency response 300 Hz	TUN = 1 kHz to 300 Hz	–0.6	+0.4	+1.4	dB
	frequency response 10 kHz	TUN = 1 kHz to 10 kHz	–9.2	–7.7	–6.2	dB
FM demodulator and noise reduction (LINEL and LINER; pins 15 and 16)						
V_o	output voltage		–10	–8	–6	dBV
α_{cb}	channel balance		–1.5	0	+1.5	dB

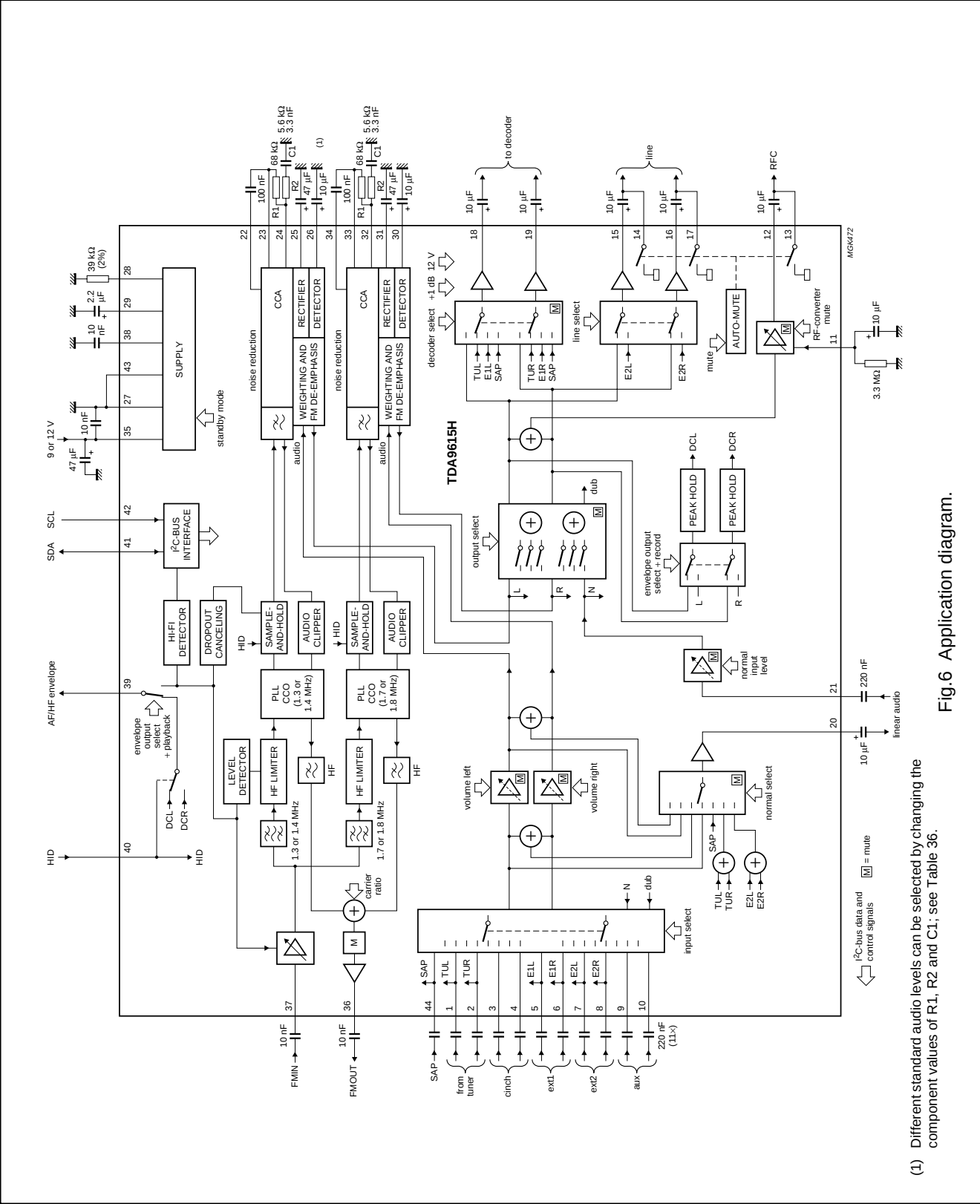
Notes

1. Signals below maximum input level are handled without internal clipping. Higher input levels however can still be handled properly by the demodulators. Typical input level equals two carriers of 30 mV (RMS value).
2. Single carrier signal at FMIN of 75 mV (RMS value) and amplitude modulated. Control bandwidth is the modulation frequency at which the amplitude modulation is attenuated 3 dB by the HF AGC.
3. Sample-and-hold audio distortion is measured using 500 Hz at HID input (pin 40). FMIN: $f_{mod} = 10 \text{ kHz}$; $\Delta f = 50 \text{ kHz}$. Audio distortion is measured using a 3 kHz 4th order low-pass filter. Value is corrected with 24 dB in order to calculate equivalent distortion at 30 Hz HID input.
4. Typical value: B = 20 Hz to 20 kHz, unweighted; production testing: B = 300 Hz to 20 kHz, unweighted.

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APPLICATION AND TEST INFORMATION



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Automatic calibration; see Fig.7

By means of bit CALS (power byte) the integrated auto-calibration system is activated. By adjusting the carrier frequencies, band-pass filters and noise reduction filters auto-calibration ensures that the hi-fi processing always is in accordance with the VHS hi-fi system standard. Calibration is only needed after start-up of the video recorder; as long as the supply voltage (pin 35) is available calibration settings remain stable.

Auto-calibration is performed with the device in hi-fi 'loop-through' mode, no standby or test mode should be active (auto-calibration demands bit settings AFM = 1, STBP = 0, STBA = 0 and TEST = 0). After setting bit CALS (power byte) to logic 1 auto-calibration is started. Calibration is performed fully automatically, using the HID input signal as a reference.

Calibration of the oscillator frequencies is performed by measuring the number of oscillator cycles within one HID HIGH period and comparing this with an internal value stored in ROM. Four different ROM values are available for NTSC or PAL (SECAM) system calibration of both the left and right channel carrier.

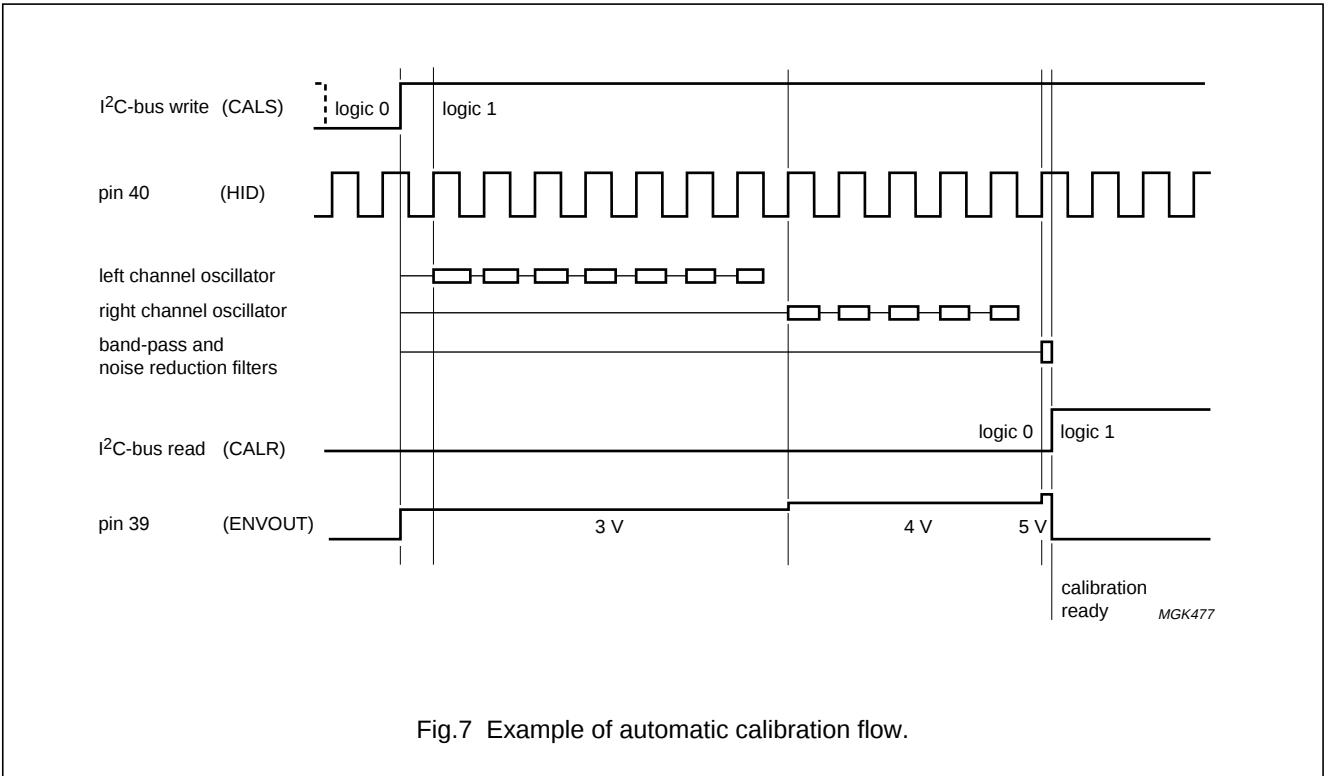
In case of NTSC calibration a special routine is active for the right channel carrier calibration resulting in a frequency difference between the left and right channel carrier of

approximately 401.2 kHz. This value effectively reduces the crosstalk from hi-fi carriers to the video colour signal as present during 'playback' mode using Extended Play (EP) tape speed.

NTSC calibration uses the standard 29.97 Hz (i.e. 16.683 ms) HID signal where PAL calibration uses the standard 25 Hz (i.e. 20 ms) HID signal. The maximum frequency error after auto-calibration is ± 5 kHz assuming a HID HIGH-time error of maximum 5 μ s. To realize NTSC EP optimization within ± 2 kHz, HID jitter should not exceed 1 μ s. In general the crystal based HID signal available in the video recorder can be used without modification.

After calibration of the oscillators, the band-pass filters are calibrated together with the integrated 'weighting' and 'FM de-emphasis' filter of the noise reduction. The total auto-calibration time needed is 17 HID cycles or less. End of calibration is signalled by bit CALR of the read byte.

The envelope output can also be used to monitor calibration; for this purpose ENVOUT (pin 39) is forced >2.5 V during calibration. The audio signal to the audio envelope (level meter) function should be muted; otherwise the audio envelope output may be >2.5 V as well making it impossible to detect end of calibration via the ENVOUT pin.



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After a calibration in 'NTSC' or 'PAL' mode the oscillators and band-pass filters can be switched between NTSC and PAL system frequencies without the need for additional calibration. Switching between these system modes is executed immediately and can be done in any operational mode. Frequency accuracy of switching is 100 kHz $\pm$ 3 kHz for both carriers. For 'record' mode however it is advised to use re-calibration after system switching; this to obtain the best possible frequency accuracy. A new auto-calibration can be started by first resetting bit CALS to logic 0 followed by setting bit CALS to logic 1 again.

carrier level (e.g. auto-tracking). At the microcontroller only one A/D converter input is needed for reading of all necessary information. During 'playback' mode I²C-bus bit EOS offers selection between audio or carrier level information. Audio level information is always output during 'loop-through' mode and 'record' mode however now EOS offers a fixed selection of hi-fi stereo (i.e. independent of the 'Output Select' mode setting); helpful when audio level information is used by the microcontroller for controlling the hi-fi record volume.

Envelope output; see Figs 8 to 11

ENVOUT (pin 39) is an analog output for stereo audio level (e.g. level meter display) and for 'playback' mode FM

Table 35 Envelope output select

AFM	EOS	ENVELOPE OUTPUT	FUNCTIONAL USE
0 (playback mode)	0	AF envelope of output select	level meter display
0 (playback mode)	1	HF envelope	auto-tracking or manual tracking display
1 (record mode/ loop-through mode)	0	AF envelope of output select	level meter display
1 (record mode/ loop-through mode)	1	AF envelope of hi-fi stereo	record volume control (and level display)

Audio envelope uses time multiplexing to output both left and right channel audio level. A peak hold function and dynamic range compression (square root function) are included for easy read-out. The peak hold function as well as left and right channel multiplexing is controlled by the HID input signal (pin 40).

The HF envelope output signal is continuous and derived from the left channel carrier. HF envelope shows a logarithmic characteristic.

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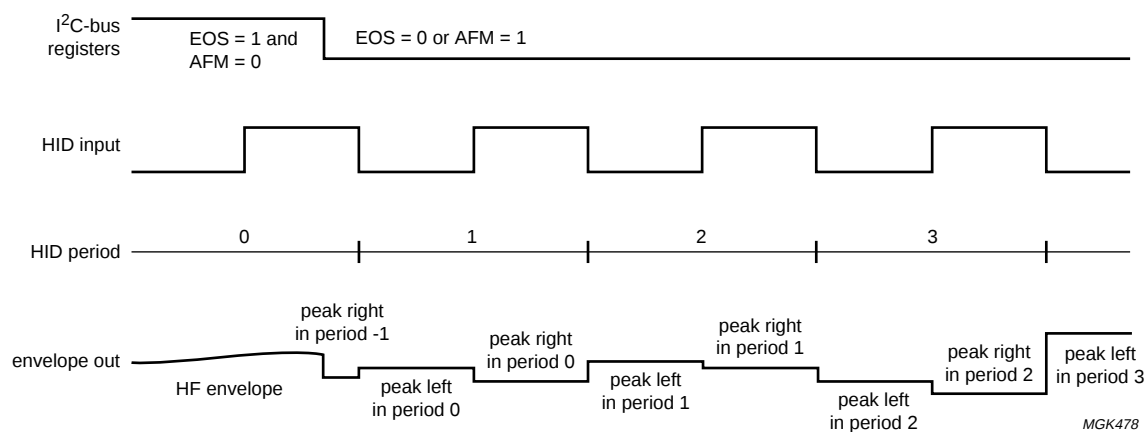


Fig.8 Timing diagram of envelope output signal.

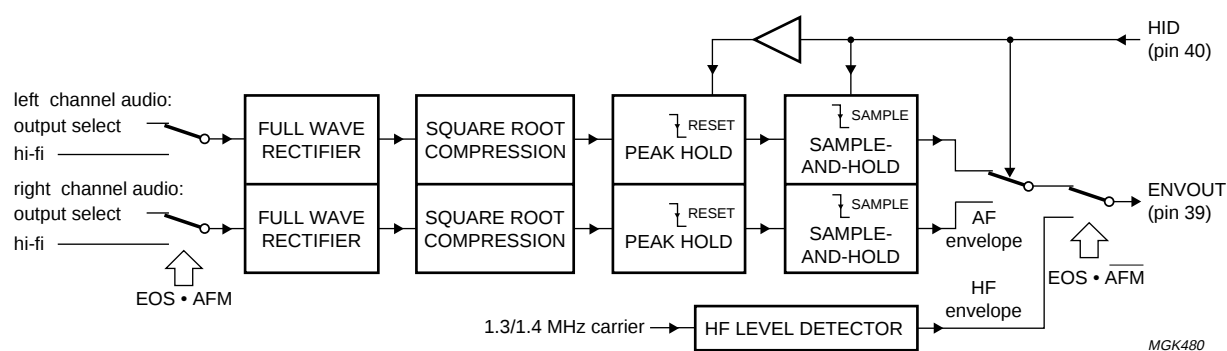
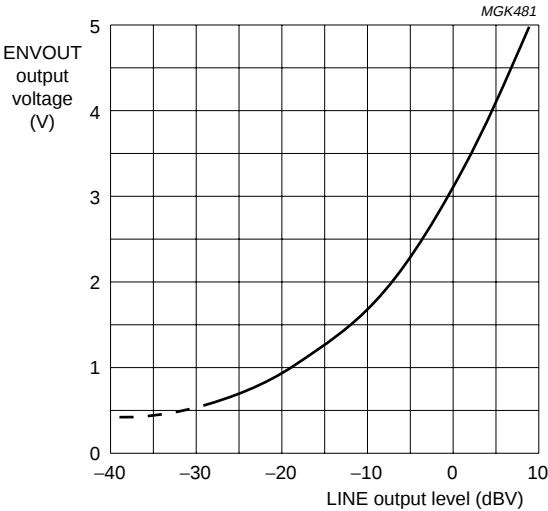


Fig.9 Functional diagram of envelope output circuit.

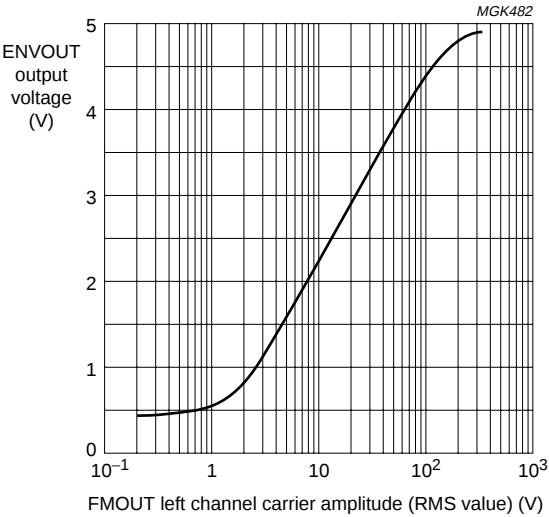
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LOH = 0.

Fig.10 Envelope output; AF envelope (audio peak level).



NTSC (1.3 MHz) or PAL (1.4 MHz).

Fig.11 Envelope output; HF envelope (playback carrier level).

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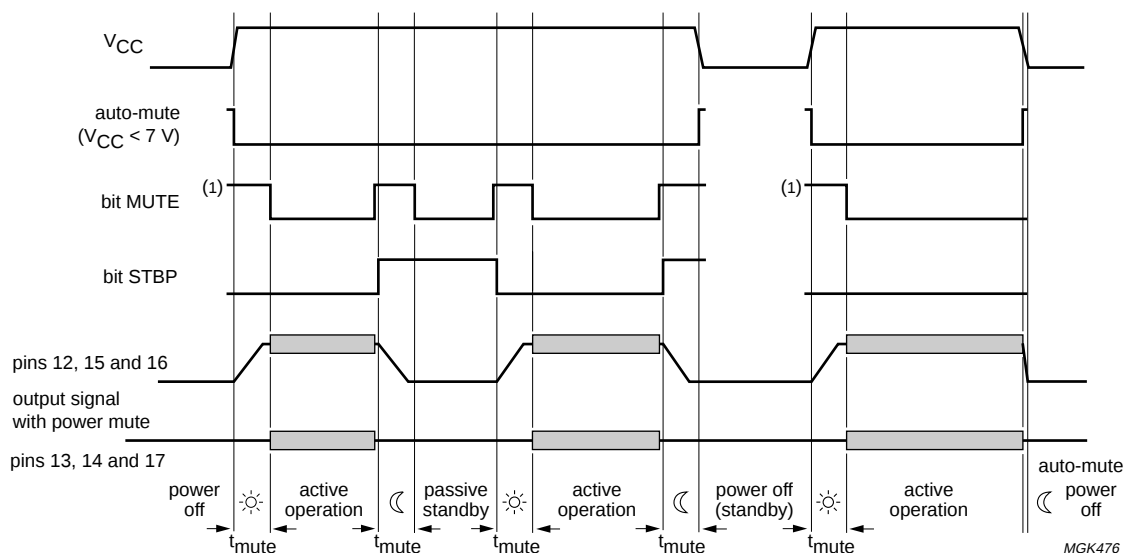
Power mute; see Fig.12

Without countermeasures switching of the power supply voltage (V_{CC} ; pin 35) or use of the built-in 'passive standby' mode causes strong disturbances on the output pins because of the rise or drop of the output's DC voltages. The TDA9615H includes three integrated mute switches to block such disturbances and avoiding the need of an external mute circuit. By connecting the power mute switches behind the line and RFC output capacitors pop-free line and RFC output signals are realized.

Power muting is active when control byte bit MUTE = 1. Because this also is the POR default state, muting is automatically activated when the power supply voltage is switched on. The integrated mute switches (MUTEC, MUTEL and MUTER; pins 13, 14 and 17) are closed, forming a low impedance path to ground. Furthermore the line and RFC outputs (RFCOUT, LINEL and LINER; pins 12, 15 and 16) are current limited to 1 mA to achieve good attenuation without the need for a series resistor between output and mute switch. Although the decoder outputs (DECL and DECR; pins 18 and 19) have no integrated muting these are current limited also for use with the integrated mute switches or to assist possible external muting.

After the output DC voltage is settled the power mute function can be deactivated by selecting MUTE = 0. Now the mute switches are opened showing a high impedance of 100 k Ω to ground and output current limiting is deactivated. Note that the time needed for output DC settling is proportional to the output capacitor value used; with a 10 μ F capacitor a safe mute time to use is 200 ms (i.e. $t_{\text{mute}} = C \times 20000$). Using such controlled muting very good performance is achieved for power-up, power-down and 'passive standby' mode switching.

At a sudden supply power loss however (e.g. mains power plug pulled) there may be no time to activate the power mute function via the I²C-bus. A power-down detector however instantly activates the power mute function in case the supply voltage falls below 7 V (auto-mute). For proper muting supply voltage drop rate should not exceed 1 V/10 ms.



(1) POR.

Fig.12 Example of power mute control and auto-mute function.

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Do not set MUTE and change STBP at the same time; to avoid output glitches power muting should precede the switching of 'passive standby' mode. Power muting takes up approximately 4 mA supply current, so to realize the minimum power consumption of 'passive standby' mode muting has to be deactivated. Note also the use of 'passive standby' mode to initiate a controlled power-off switching.

I_{ref} resistor

The external resistor at pin 28 defines internal reference currents and determines temperature stability of the circuits adjusted by the auto-calibration function.

The external resistor should have the following dimensions: value = 39.0 kΩ ±2%; temperature coefficient = ±50 ppm/K.

Standard hi-fi audio level

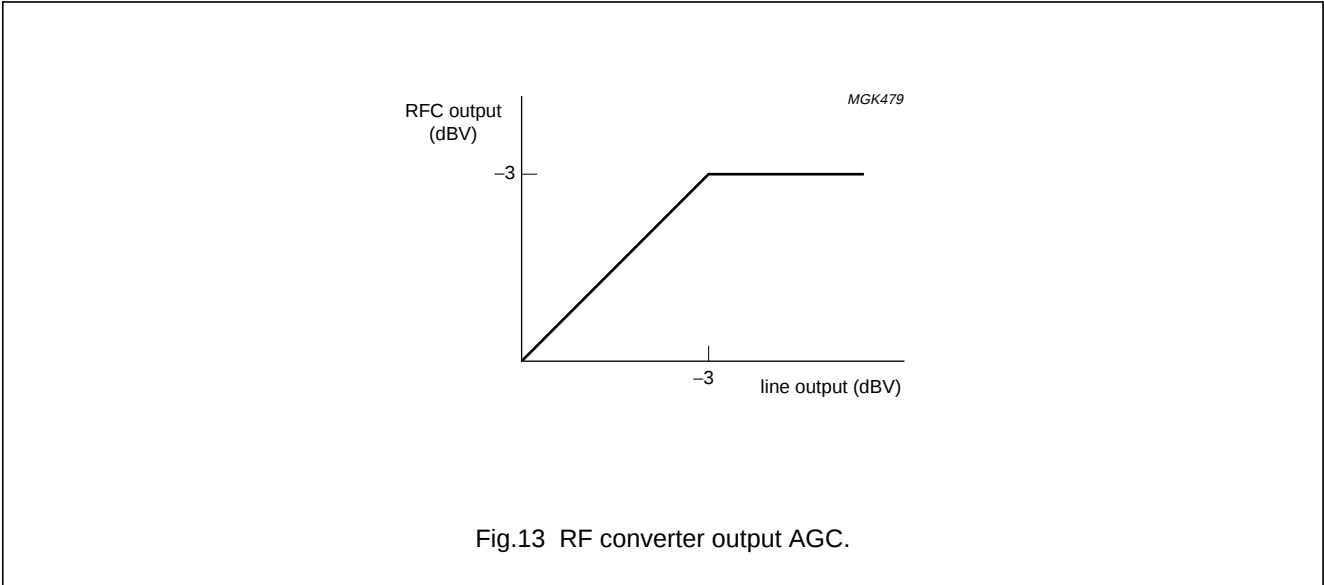
Using the application circuit as proposed in Fig.6, the standard FM deviation of 50 kHz equals a 1 kHz audio signal of -8 dBV line output level (LOH = 0). A different standard audio level can be selected by changing the external filter components of the noise reduction at pins 24 and 32 (EMPHL and EMPHR); standard audio level changes proportional to the impedance of the external de-emphasis filter; see Table 36.

Table 36 Standard audio levels; see application diagram of Fig.6

COMPONENT VALUES			LINE OUTPUT LEVEL FOR 50 kHz FM DEVIATION (EQUIVALENT TO 1 kHz AUDIO SIGNAL) (dBV)
R1 (kΩ)	R2 (kΩ)	C1 (nF)	
91	7.5	2.4	-5.6
82	6.8	2.7	-6.4
75	6.2	3.0	-7.1
68	5.6	3.3	-8.0
62	5.1	3.6	-8.8
56	4.7	3.9	-9.8
51	4.3	4.3	-10.6

RFC AGC; see Fig.13

To avoid over modulation in the RF converter unit connected to RFCOUT (pin 12) an AGC function is incorporated, limiting the maximum signal level at pin RFCOUT to -3 dBV.



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The RFC output can be muted by setting of bit RFCM = 1. When applying this mute the AGC control is reset by discharging the capacitor at pin 11 (RFCAGC).

Audio dubbing

TDA9615H includes unparalleled functionality supporting the audio dubbing function of hi-fi video recorders. Audio dubbing is a feature mode for recording of new sound material on the linear audio track (i.e. normal sound) of an existing recording. Audio dubbing can be used in two different ways.

OUTPUT MIX

A new, partly, recording is made on the linear audio track; afterwards during 'playback' mode the new linear audio and the original hi-fi sound are combined. This way hi-fi stereo quality remains and linear audio is only partly used e.g. for speech only (commentary track). However there is no control over the original (hi-fi) sound.

'Playback' mode mixing of hi-fi and normal sound is supported by the output select function modes 'Mix-Left', 'Mix-Right' and 'Mix-Stereo' (OSN, OSR and OSL of the output byte) creating a fixed output signal of $\frac{1}{2}$ hi-fi left + $\frac{1}{2}$ normal.

INPUT MIX

A new, complete, recording is made on the linear audio track; afterwards 'playback' mode uses linear audio sound only. This way hi-fi stereo quality is lost but there is total freedom in defining the new material. Furthermore this way 'playback' mode is no longer restricted to hi-fi video recorders with output mix option.

Selection 'Dub Mix' mode of the input select function, when combined with selection 'Volume' mode of the normal select function, changes the circuit into a mixing desk. A new linear audio recording can be created mixing together new and original sound. By use of the left and right channel volume controls continuous control is offered over amplitude and ratio mix of the AUX input signal (e.g. a microphone input) and the original hi-fi 'playback' mode sound.

This functionality is realized by internal connection of the AUX input pair to the left channel volume control and by internal connection of the hi-fi output signal pair to the right channel volume control. For output and hi-fi selection the 'Dub Mix' mode uses, and partly overrules, the output select function settings. Output select function modes 'Mix-Left', 'Mix-Right' and 'Mix-Stereo' make the normal sound available at the line outputs for monitoring of the 'Dub Mix' mode recording.

Table 37 Input select function is set to 'Dub Mix' mode (IS2, IS1, IS0 = 101); note 1

OUTPUT SELECT MODE	OSN	OSR	OSL	DUB MIX OUTPUT SELECTION	DUB MIX INPUT SELECTION	
					LEFT CHANNEL	RIGHT CHANNEL
Mute	0	0	0	mute	aux stereo	mute
Left	0	0	1	hi-fi left	aux stereo	hi-fi left
Right	0	1	0	hi-fi right	aux stereo	hi-fi right
Stereo	0	1	1	stereo	aux stereo	stereo
Normal	1	0	0	normal	aux stereo	normal
Mix-Left	1	0	1	normal	aux stereo	hi-fi left
Mix-Right	1	1	0	normal	aux stereo	hi-fi right
Mix-Stereo	1	1	1	normal	aux stereo	hi-fi stereo

Note

1. Modes shown in bold are the most used modes.

Output select function mode 'Mix-Stereo' is the setting generally used for audio dubbing. In combination with setting the normal select function to 'Volume' mode user control over amplitude and ratio is offered for the AUX and hi-fi signal as follows: $(\frac{1}{4}\text{AUXL} + \frac{1}{4}\text{AUXR}) \times \text{volume left} + (\frac{1}{4}\text{hi-fi left} + \frac{1}{4}\text{hi-fi right}) \times \text{volume right}$.

Switching hi-fi to a mode other than 'playback' or selection of output select function mode 'Normal' should be avoided; using these settings a signal loop can be closed from output to input possibly leading to audio oscillation. For the same reason auto-normal switching is not active during 'Dub Mix' mode; in case no hi-fi input signal is detected the hi-fi sound is muted.

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Test modes; see Table 38

Special test modes are implemented for testing and evaluation purposes. These test modes are available via the power byte TEST bit and selection is enabled via bits of the select byte.

Table 38 Test modes for evaluation purposes

TEST	s4	NIL3	NIL2	NIL1	NIL0	DESCRIPTION
0	X ⁽¹⁾	X ⁽¹⁾	X ⁽¹⁾	X ⁽¹⁾	X ⁽¹⁾	standard operation
1	0	0	0	0	1	test 1: left channel FM carrier only (1.3 or 1.4 MHz); record mode
1	0	0	0	1	0	test 2: right channel FM carrier only (1.7 or 1.8 MHz); record mode
1	0	0	0	1	1	test 3: left channel band-pass filter, HF AGC off; playback mode (test output = FMOUT)
1	0	0	1	0	0	test 4: right channel band-pass filter, HF AGC off; playback mode (test output = FMOUT)
1	0	0	1	0	1	test 5: HF AGC (via left channel band-pass filter); playback mode (test output = FMOUT)
1	0	0	1	1	0	test 6: HF AGC (via right channel band-pass filter); playback mode (test output = FMOUT)
100111 to 111000						product testing; not for evaluation; note 2
1	1	1	0	0	1	test 25 (note 3) a: left channel FM modulator (left carrier only); record mode (test input = line in left) b: left and right channel noise reduction (compressor); record mode (test output = line out) c: left and right channel FM demodulator; playback mode (test output = line out)
1	1	1	0	1	0	test 26 (note 3) a: right channel FM modulator (right carrier only); record mode (test input = line in right) b: left and right channel audio low-pass filter; record mode (test input = line in, test output = line out) c: left and right channel noise reduction (expander); playback mode (test input = line in); note 4

Notes

1. X = don't care.
2. Calibration may be lost when selecting product testing modes.
3. Audio output level for tests 25b and 25c (test output = line out) is typical –6.5 dBV (with input = –8 dBV or FM deviation = 50 kHz; LOH = 0). Audio input level for tests 25a, 26a and 26c (test input = line in) is typical –6.5 dBV (for output = –8 dBV or FM deviation = 50 kHz; hi-fi volume = –3 dB). The test outputs and test inputs used for tests 25 and 26 directly connect to internal signal lines. Signals found here are not compensated for temperature or tolerance spread, level measurements therefore can be used relative only. Absolute values are no indication of overall performance.
4. Test 26c (expander test) requires the auto-normal function to be deactivated (i.e. carrier available at FMIN pin).

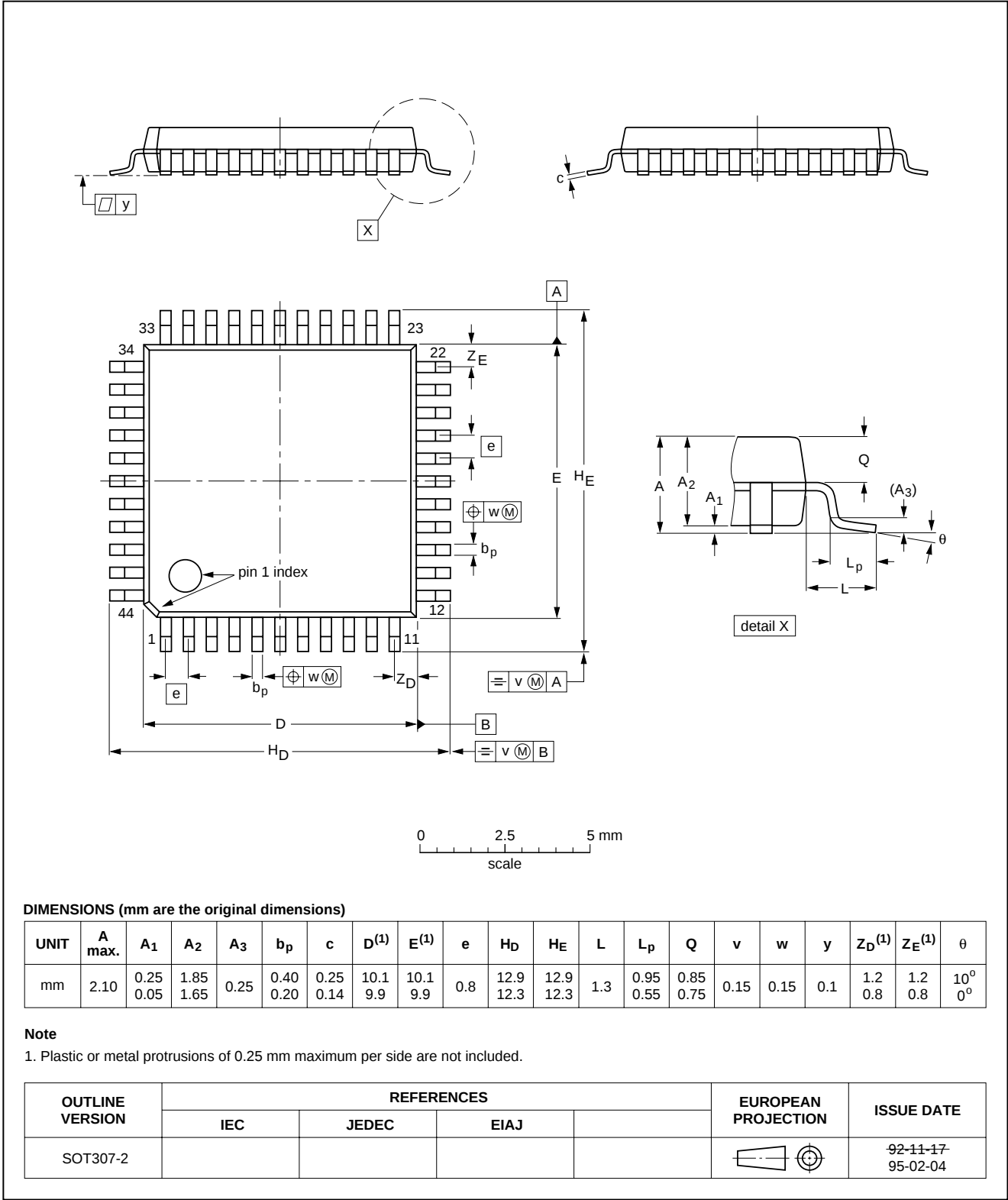
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PACKAGE OUTLINE

QFP44: plastic quad flat package; 44 leads (lead length 1.3 mm); body 10 x 10 x 1.75 mm

SOT307-2



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SOLDERING**Introduction**

There is no soldering method that is ideal for all IC packages. Wave soldering is often preferred when through-hole and surface mounted components are mixed on one printed-circuit board. However, wave soldering is not always suitable for surface mounted ICs, or for printed-circuits with high population densities. In these situations reflow soldering is often used.

This text gives a very brief insight to a complex technology. A more in-depth account of soldering ICs can be found in our *"IC Package Databook"* (order code 9398 652 90011).

Reflow soldering

Reflow soldering techniques are suitable for all QFP packages.

The choice of heating method may be influenced by larger plastic QFP packages (44 leads, or more). If infrared or vapour phase heating is used and the large packages are not absolutely dry (less than 0.1% moisture content by weight), vaporization of the small amount of moisture in them can cause cracking of the plastic body. For more information, refer to the Drypack chapter in our *"Quality Reference Handbook"* (order code 9397 750 00192).

Reflow soldering requires solder paste (a suspension of fine solder particles, flux and binding agent) to be applied to the printed-circuit board by screen printing, stencilling or pressure-syringe dispensing before package placement.

Several techniques exist for reflowing; for example, thermal conduction by heated belt. Dwell times vary between 50 and 300 seconds depending on heating method. Typical reflow temperatures range from 215 to 250 °C.

Preheating is necessary to dry the paste and evaporate the binding agent. Preheating duration: 45 minutes at 45 °C.

Wave soldering

Wave soldering is **not** recommended for QFP packages. This is because of the likelihood of solder bridging due to closely-spaced leads and the possibility of incomplete solder penetration in multi-lead devices.

If wave soldering cannot be avoided, the following conditions must be observed:

- **A double-wave (a turbulent wave with high upward pressure followed by a smooth laminar wave) soldering technique should be used.**
- **The footprint must be at an angle of 45° to the board direction and must incorporate solder thieves downstream and at the side corners.**

Even with these conditions, do not consider wave soldering the following packages: QFP52 (SOT379-1), QFP100 (SOT317-1), QFP100 (SOT317-2), QFP100 (SOT382-1) or QFP160 (SOT322-1).

During placement and before soldering, the package must be fixed with a droplet of adhesive. The adhesive can be applied by screen printing, pin transfer or syringe dispensing. The package can be soldered after the adhesive is cured.

Maximum permissible solder temperature is 260 °C, and maximum duration of package immersion in solder is 10 seconds, if cooled to less than 150 °C within 6 seconds. Typical dwell time is 4 seconds at 250 °C.

A mildly-activated flux will eliminate the need for removal of corrosive residues in most applications.

Repairing soldered joints

Fix the component by first soldering two diagonally-opposite end leads. Use only a low voltage soldering iron (less than 24 V) applied to the flat part of the lead. Contact time must be limited to 10 seconds at up to 300 °C. When using a dedicated tool, all other leads can be soldered in one operation within 2 to 5 seconds between 270 and 320 °C.

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DEFINITIONS

Data sheet status	
Objective specification	This data sheet contains target or goal specifications for product development.
Preliminary specification	This data sheet contains preliminary data; supplementary data may be published later.
Product specification	This data sheet contains final product specifications.
Limiting values	
Limiting values given are in accordance with the Absolute Maximum Rating System (IEC 134). Stress above one or more of the limiting values may cause permanent damage to the device. These are stress ratings only and operation of the device at these or at any other conditions above those given in the Characteristics sections of the specification is not implied. Exposure to limiting values for extended periods may affect device reliability.	
Application information	
Where application information is given, it is advisory and does not form part of the specification.	

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NOTES

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