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DATA SHEET

TDA9875A
Digital TV Sound Processor (DTVSP)

Product specification
Supersedes data of 1998 Aug 13
File under Integrated Circuits, IC02

1999 Dec 20

Philips
Semiconductors

PHILIPS

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1.1 Demodulator and decoder section

- Sound IF (SIF) input switch e.g. to select between terrestrial TV SIF and SAT SIF sources
- SIF AGC with 24 dB control range
- SIF 8-bit Analog-to-Digital Converter (ADC)
- Differential Quadrature Phase Shift Keying (DQPSK) demodulation for different standards, simultaneously with 1-channel FM demodulation
- Near Instantaneous Companded Audio Multiplex (NICAM) decoding (B/G, I and L standard)
- Two-carrier multistandard FM demodulation (B/G, D/K and M standard)
- Decoding for three analog multi-channel systems (A2, A2+ and A2*) and satellite sound
- Optional AM demodulation for system L, simultaneously with NICAM
- Programmable identification (B/G, D/K and M standard) and different identification times.

1.2 DSP section

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- Control of volume, balance, contour, bass, treble, pseudo stereo, spatial, bass boost and soft mute
- Plop-free volume control
- Automatic Volume Level (AVL) control
- Adaptive de-emphasis for satellite
- Programmable beeper
- Monitor selection for FM/AM DC values and signals, with peak detection option
- I²S-bus interface for a feature extension (e.g. Dolby Pro Logic) with matrix, level adjust and mute.

1.3 Analog audio section

- Analog crossbar switch with inputs for mono and stereo (also applicable as SCART 3 input), SCART 1 input/output, SCART 2 input/output and line output
- User defined full-level/–3 dB scaling for SCART outputs
- Output selection of mono, stereo, dual A/B, dual A or dual B
- 20 kHz bandwidth for SCART-to-SCART copies
- Standby mode with function for SCART copies

- Dual audio Digital-to-Analog Converter (DAC) from DSP to analog crossbar switch, bandwidth 15 kHz
- Dual audio ADC from analog inputs to DSP
- Two dual audio DACs for loudspeaker (Main) and headphone (Auxiliary) outputs; also applicable for L, R, C and S in the Dolby Pro Logic mode with feature extension.

2 GENERAL DESCRIPTION

The TDA9875A is a single-chip Digital TV Sound Processor (DTVSP) for analog and digital multi-channel sound systems in TV sets and satellite receivers.

2.1 Supported standards

The multistandard/multi-stereo capability of the TDA9875A is mainly of interest in Europe, but also in Hong Kong/Peoples Republic of China and South East Asia. This includes B/G, D/K, I, M and L standards. In other application areas there exists only subsets of these standard combinations otherwise only single standards are transmitted.

M standard is transmitted in Europe by the American Forces Network (AFN) with European channel spacing (7 MHz VHF and 8 MHz UHF) and monaural sound.

The AM sound of L/L accent standard is normally demodulated in the first sound IF. The resulting AF signal has to be entered into the mono audio input of the TDA9875A. A second possibility is to use the internal AM demodulator stage, however this gives limited performance.

Korea has a stereo sound system similar to Europe and is supported by the TDA9875A. The differences include deviation, modulation contents and identification. It is based on M standard.

An overview of the supported standards and sound systems and their key parameters is given in Table 1.

The analog multi-channel sound systems (A2, A2+ and A2*) are 2-Carrier Systems (2CS).

2.1.1 ANALOG 2-CARRIER SYSTEMS

Table 1 Frequency modulation

STANDARD	SOUND SYSTEM	CARRIER FREQUENCY (MHz)	FM DEVIATION (kHz)			MODULATION		BANDWIDTH/ DE-EMPHASIS (kHz/μs)
			NOM.	MAX.	OVER	SC1	SC2	
M	mono	4.5	15	25	50	mono	–	15/75
M	A2+	4.5/4.724	15	25	50	½(L + R)	½(L – R)	15/75 (Korea)
B/G	A2	5.5/5.742	27	50	80	½(L + R)	R	15/50
I	mono	6.0	27	50	80	mono	–	15/50
D/K	A2	6.5/6.742	27	50	80	½(L + R)	R	15/50
D/K	A2*	6.5/6.258	27	50	80	½(L + R)	R	15/50

Table 2 Identification for A2 systems

PARAMETER	A2/A2*	A2+ (KOREA)
Pilot frequency	54.6875 kHz = 3.5 × line frequency	55.0699 kHz = 3.5 × line frequency
Stereo identification frequency	$117.5\ \text{Hz} = \frac{\text{line frequency}}{133}$	$149.9\ \text{Hz} = \frac{\text{line frequency}}{105}$
Dual identification frequency	$274.1\ \text{Hz} = \frac{\text{line frequency}}{57}$	$276.0\ \text{Hz} = \frac{\text{line frequency}}{57}$
AM modulation depth	50%	50%

2.1.2 2-CARRIER SYSTEMS WITH NICAM

Table 3 NICAM

STANDARD	SC1						SC2 NICAM (MHz)	DE-EMPHASIS	ROLL-OFF (%)	NICAM CODING
	FREQUENCY (MHz)	TYPE	MODULATION							
			INDEX (%)		DEVIATION (kHz)					
			NOM.	MAX.	NOM.	MAX.				
B/G	5.5	FM	–	–	27	50	5.85	J17	40	note 1
I	6.0	FM	–	–	27	50	6.552	J17	100	note 1
D/K	6.5	FM	–	–	27	50	5.85	J17	40	note 2
L	6.5	AM	54	100	–	–	5.85	J17	40	note 1

Notes

1. See *"EBU specification"* or equivalent specification.
2. Not yet defined.

2.1.3 SATELLITE SYSTEMS

An important specification for satellite TV reception is the *'Astra specification'*. The TDA9875A is suited for the reception of Astra and other satellite signals.

Table 4 FM satellite sound

CARRIER TYPE	CARRIER FREQUENCY (MHz)	MODULATION INDEX	MAXIMUM FM DEVIATION (kHz)	MODULATION	BANDWIDTH/ DE-EMPHASIS (kHz/μs)
Main	6.50[1]	0.26	85	mono	15/50[2]
Sub	7.02/7.20	0.15	50	m/st/d[3]	15/adaptive[4]
Sub	7.38/7.56				
Sub	7.74/7.92				
Sub	8.10/8.28				

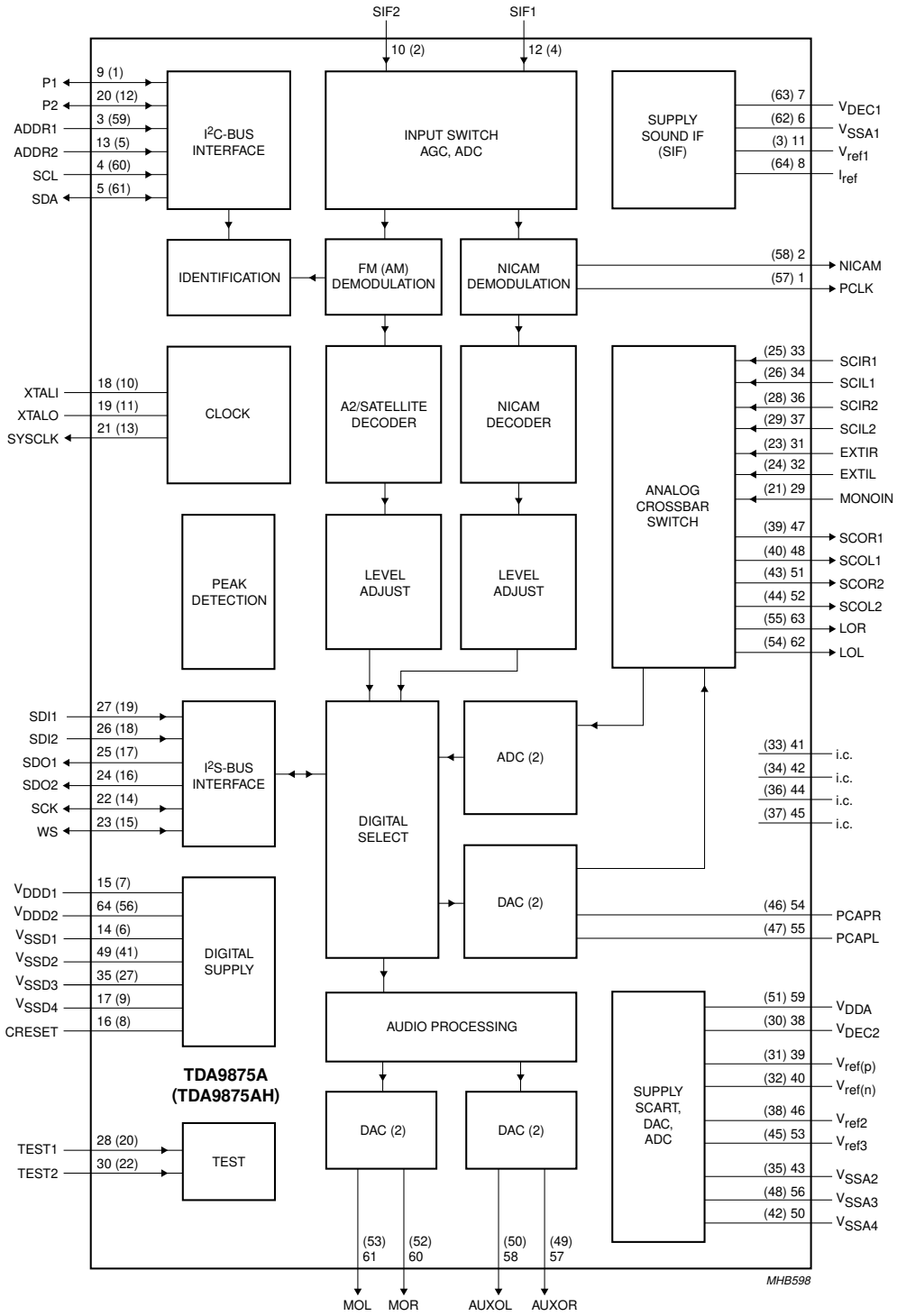
Notes

1. For other satellite systems, frequencies of e.g. 5.80, 6.60 or 6.65 MHz can also be received.
2. A de-emphasis of 60 μs, or in accordance with J17, is available.
3. m/st/d = mono, stereo or dual language sound.
4. Adaptive de-emphasis is compatible to transmitter specification.

3 ORDERING INFORMATION

TYPE NUMBER	PACKAGE		
	NAME	DESCRIPTION	VERSION
TDA9875A	SDIP64	plastic shrink dual in-line package; 64 leads (750 mil)	SOT274-1
TDA9875AH	QFP64	plastic quad flat package; 64 leads (lead length 1.6 mm); body $14 \times 14 \times 2.7$ mm	SOT393-1

4 BLOCK DIAGRAM



The pin numbers given in parenthesis refer to the TDA9875AH version.

Fig.1 Block diagram.

5 PINNING

SYMBOL	PIN		PIN TYPE(1)	DESCRIPTION
	TDA9875A	TDA9875AH		
PCLK	1	57	O	NICAM clock output at 728 kHz
NICAM	2	58	O	serial NICAM data output at 728 kHz
ADDR1	3	59	I	I^2C-bus slave address input 1
SCL	4	60	I	I^2C-bus clock input
SDA	5	61	I/O	I^2C-bus data input/output
V_{SSA1}	6	62	S	supply ground 1; analog front-end circuitry
V_{DEC1}	7	63	–	supply voltage decoupling 1; analog front-end circuitry
I_{ref}	8	64	–	resistor for reference current generator; analog front-end circuitry
P1	9	1	I/O	general purpose input/output pin 1
SIF2	10	2	I	sound IF input 2
V_{ref1}	11	3	–	reference voltage 1; analog front-end circuitry
SIF1	12	4	I	sound IF input 1
ADDR2	13	5	I	I^2C-bus slave address input 2
V_{SSD1}	14	6	S	supply ground 1; digital circuitry
V_{DDD1}	15	7	S	digital supply voltage 1; digital circuitry
CRESET	16	8	–	capacitor for Power-on reset
V_{SSD4}	17	9	S	supply ground 4; digital circuitry
XTALI	18	10	I	crystal oscillator input
XTALO	19	11	O	crystal oscillator output
P2	20	12	I/O	general purpose input/output pin 2
SYSCLK	21	13	O	system clock output
SCK	22	14	I/O	I^2S-bus clock input/output
WS	23	15	I/O	I^2S-bus word select input/output
SDO2	24	16	O	I^2S-bus data output 2 (I^2S2 output)
SDO1	25	17	O	I^2S-bus data output 1 (I^2S1 output)
SDI2	26	18	I	I^2S-bus data input 2 (I^2S2 input)
SDI1	27	19	I	I^2S-bus data input 1 (I^2S1 input)
TEST1	28	20	I	test pin 1; connected to V_{SSD1} for normal operating mode
MONOIN	29	21	I	audio mono input
TEST2	30	22	I	test pin 2; connected to V_{SSD1} for normal operating mode
EXTIR	31	23	I	external audio input right channel
EXTIL	32	24	I	external audio input left channel
SCIR1	33	25	I	SCART 1 input right channel
SCIL1	34	26	I	SCART 1 input left channel
V_{SSD3}	35	27	S	supply ground 3; digital circuitry
SCIR2	36	28	I	SCART 2 input right channel
SCIL2	37	29	I	SCART 2 input left channel
V_{DEC2}	38	30	–	supply voltage decoupling 2; audio analog-to-digital converter circuitry

SYMBOL	PIN		PIN TYPE(1)	DESCRIPTION
	TDA9875A	TDA9875AH		
$V_{ref(p)}$	39	31	–	positive reference voltage; audio analog-to-digital converter circuitry
$V_{ref(n)}$	40	32	–	reference voltage ground; audio analog-to-digital converter circuitry
i.c.	41	33	–	internally connected; note 2
i.c.	42	34	–	internally connected; note 3
V_{SSA2}	43	35	S	supply ground 2; audio analog-to-digital converter circuitry
i.c.	44	36	–	internally connected; note 3
i.c.	45	37	–	internally connected; note 2
V_{ref2}	46	38	–	reference voltage 2; audio analog-to-digital converter circuitry
SCOR1	47	39	O	SCART 1 output right channel
SCOL1	48	40	O	SCART 1 output left channel
V_{SSD2}	49	41	S	supply ground 2; digital circuitry
V_{SSA4}	50	42	S	supply ground 4; audio operational amplifier circuitry
SCOR2	51	43	O	SCART 2 output right channel
SCOL2	52	44	O	SCART 2 output left channel
V_{ref3}	53	45	–	reference voltage 3; audio digital-to-analog converter and operational amplifier circuitry
PCAPR	54	46	–	post-filter capacitor pin right channel; audio digital-to-analog converter
PCAPL	55	47	–	post-filter capacitor pin left channel; audio digital-to-analog converter
V_{SSA3}	56	48	S	supply ground 3; audio digital-to-analog converter circuitry
AUXOR	57	49	O	headphone (Auxiliary) output right channel
AUXOL	58	50	O	headphone (Auxiliary) output left channel
V_{DDA}	59	51	S	analog supply voltage; analog circuitry
MOR	60	52	O	loudspeaker (Main) output right channel
MOL	61	53	O	loudspeaker (Main) output left channel
LOL	62	54	O	line output left channel
LOR	63	55	O	line output right channel
V_{DDD2}	64	56	S	digital supply voltage 2; digital circuitry

Notes

1. Pin type: I = input, O = output, S = supply.
2. Test pin: CMOS level input; pull-up resistor; can be connected to V_{SS}.
3. Test pin: CMOS 3-state stage; can be connected to V_{SS}.

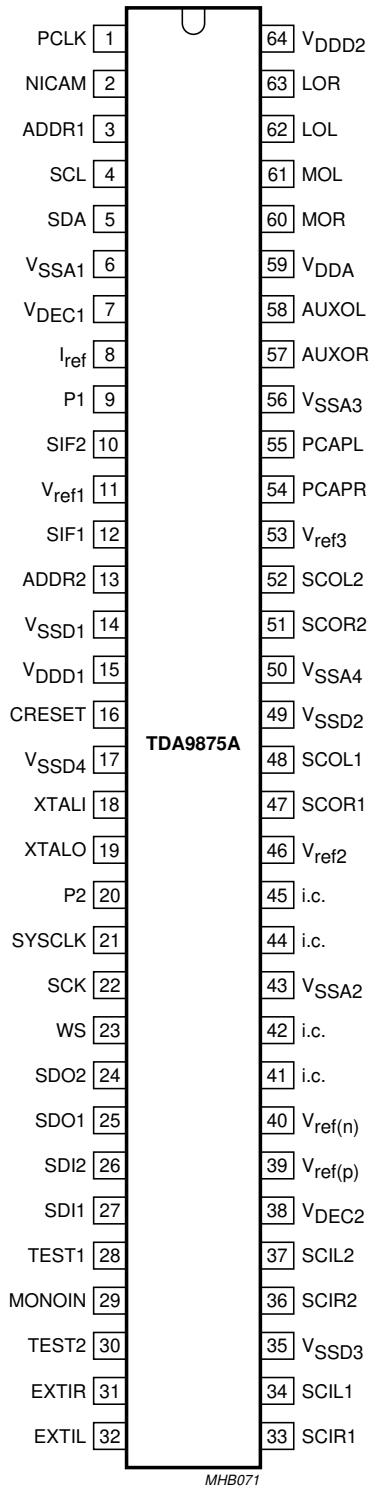


Fig.2 Pin configuration (TDA9875A).

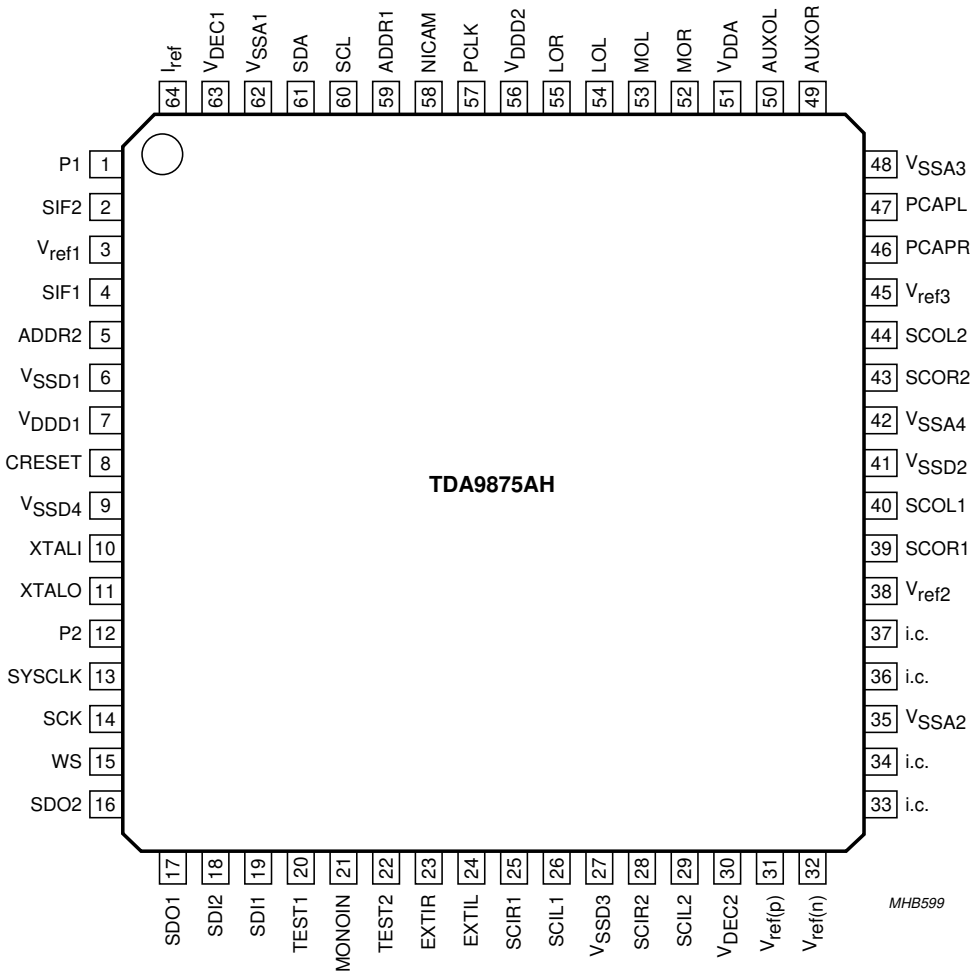


Fig.3 Pin configuration (TDA9875AH).

6 FUNCTIONAL DESCRIPTION

6.1 Demodulator and decoder section

6.1.1 SIF INPUT

Two input pins are provided: SIF1 e.g. for terrestrial TV and SIF2 e.g. for a satellite tuner. For higher SIF signal levels the SIF input can be attenuated with an internal switchable −10 dB resistor divider. As no specific filters are integrated, both inputs have the same specification giving flexibility in application. The selected signal is passed through an AGC circuit and then digitized by an 8-bit ADC operating at 24.576 MHz.

6.1.2 AGC

The gain of the AGC amplifier is controlled from the ADC output by means of a digital control loop employing hysteresis. The AGC has a fast attack behaviour to prevent ADC overloads and a slow decay behaviour to prevent AGC oscillations. For AM demodulation the AGC must be switched off. When switched off, the control loop is reset and fixed gain settings can be chosen (see Table 15).

The AGC can be controlled via the I²C-bus. Details can be found in the I²C-bus register definitions (see Chapter 10).

6.1.3 MIXER

The digitized input signal is fed to the mixers, which mix one or both input sound carriers down to zero IF. A 24-bit control word for each carrier sets the required frequency. Access to the mixer control word registers is via the I²C-bus. When receiving NICAM programs, a feedback signal is added to the control word of the second carrier mixer to establish a carrier-frequency loop.

6.1.4 FM AND AM DEMODULATION

An FM or AM input signal is fed via a band-limiting filter to a demodulator that can be used for either FM or AM demodulation. Apart from the standard (fixed) de-emphasis characteristic, an adaptive de-emphasis is available for encoded satellite programs. A stereo decoder recovers the left and right signal channels from the demodulated sound carriers. Both the European and Korean stereo systems are supported.

6.1.5 FM IDENTIFICATION

The identification of the FM sound mode is performed by AM synchronous demodulation of the pilot signal and narrow-band detection of the identification frequencies. The result is available via the I²C-bus interface. A selection can be made via the I²C-bus for B/G, D/K and M standard and for three different modes that represent different trade-offs between speed and reliability of identification.

6.1.6 NICAM DEMODULATION

The NICAM signal is transmitted in a DQPSK code at a bit rate of 728 kbit/s. The NICAM demodulator performs DQPSK demodulation and feeds the resulting bitstream and clock signal onto the NICAM decoder and, for evaluation purposes, to pins PCLK and NICAM.

A timing loop controls the frequency of the crystal oscillator to lock the sampling rate to the symbol timing of the NICAM data.

6.1.7 NICAM DECODER

The device performs all decoding functions in accordance with the *“EBU NICAM 728 specification”*. After locking to the frame alignment word, the data is descrambled by applying the defined pseudo-random binary sequence and the device will then synchronize to the periodic frame flag bit C0.

Bit VDSP (see Section 10.4.1) indicates that the decoder has locked to the NICAM data and that the data is valid sound data.

The status of the NICAM decoder can be read out from the NICAM status register by the user (see Section 10.4.2). Bit OSB indicates that the decoder has locked to the NICAM data. Bit C4 indicates that the sound conveyed by the FM mono channel is identical to the sound signal conveyed by the NICAM channel.

The error byte contains the number of sound sample errors, resulting from parity checking, that occurred in the past 128 ms period. The Bit Error Rate (BER) can be calculated using the following equation:

$$\mathrm{BER} = \frac{\text{bit errors}}{\text{total bits}} \approx \text{error byte} \times 1.74 \times 10^{-5}$$

6.1.8 NICAM AUTO-MUTE

This function is enabled by setting bit AMUTE to logic 0 (see Section 10.3.11).

Upper and lower error limits may be defined by writing appropriate values to two registers in the I²C-bus section (see Sections 10.3.13 and 10.3.14). When the number of errors in a 128 ms period exceeds the upper error limit the auto-mute function will switch the output sound from NICAM to whatever sound is on the first sound carrier (FM or AM). When the error count is smaller than the lower error limit the NICAM sound is restored.

The auto-mute function can be disabled by setting bit AMUTE to logic 1. In this condition clicks become audible when the error count increases; the user will hear a signal of degrading quality.

A decision to enable/disable the auto-muting is taken by the microcontroller based on an interpretation of the application control bits C1, C2, C3 and C4 and, possibly, any additional strategy implemented by the set maker in the microcontroller software.

For NICAM L applications, it is recommended to demodulate AM sound in the first sound IF and connect the audio signal to the mono input of the TDA9875A. By setting bit AMSEL (see Section 10.3.11), the auto-mute function will switch to the audio ADC instead of switching to the first sound carrier. The ADC source selector (see Section 10.3.20) should be set to mono input, where the AM sound signal should be connected.

6.1.9 CRYSTAL OSCILLATOR

The circuitry of the crystal oscillator is fully integrated, only the external 24.576 MHz crystal is needed (see Fig.10).

6.1.10 TEST PINS

Test pins TEST1 and TEST2 are active HIGH and in the normal operating mode of the device they are connected to V_{SSD1}. Test functions are for manufacturing tests only and are not available to customers. Without external circuitry these pins are pulled down to a LOW level with internal resistors.

6.1.11 POWER FAIL DETECTOR

The power fail detector monitors the internal power supply for the digital part of the device. If the supply has temporarily been lower than the specified lower limit, the Power-on reset bit POR (see Section 10.4.1), will be set to logic 1.

Bit CLRPOR (see Section 10.3.2) resets the Power-on reset flip-flop to LOW. If this is detected, an initialization of the TDA9875A has to be carried out to ensure reliable operation.

6.1.12 POWER-ON RESET

The reset is active LOW. In order to perform a reset at power-up, a simple RC circuit may be used which consists of the integrated passive pull-up resistor and an external capacitor connected to ground. The pull-up resistor has a nominal value of 50 kΩ, which can easily be measured between pins CRESET and V_{DDD2}. Before the supply voltage has reached a certain minimum, the state of the circuit is completely undefined, and it remains in this undefined state unless a reset is applied.

The reset is guaranteed to be active when:

- The power supply is within the specified limits (4.75 and 5.5 V)
- The crystal oscillator is functioning
- The voltage at pin CRESET is below $0.3V_{DDD}$ (1.5 V if V_{DDD} = 5.0 V, typically below 1.8 V).

The required capacitor value depends on the gradient of the rising power supply voltage. The time constant of the RC circuit should be clearly larger than the rise time of the power supply, to make sure that the reset condition is always satisfied (see Fig.4), even considering the tolerance spread. To avoid problems with a too slow discharging of the capacitor at power-down, it may be helpful to add a diode from pin CRESET to V_{DDD}. It should be noted that the internal ESD protection diode does not help here as it only conducts at higher voltages. Under difficult power supply conditions (e.g. very slow or non-monotonic ramp-up), it is recommended to drive the reset line from a microcontroller port or the like.

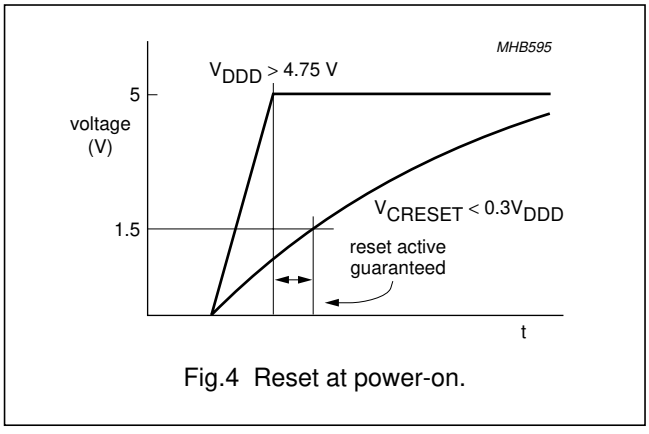


Fig.4 Reset at power-on.

6.2 Digital signal processing

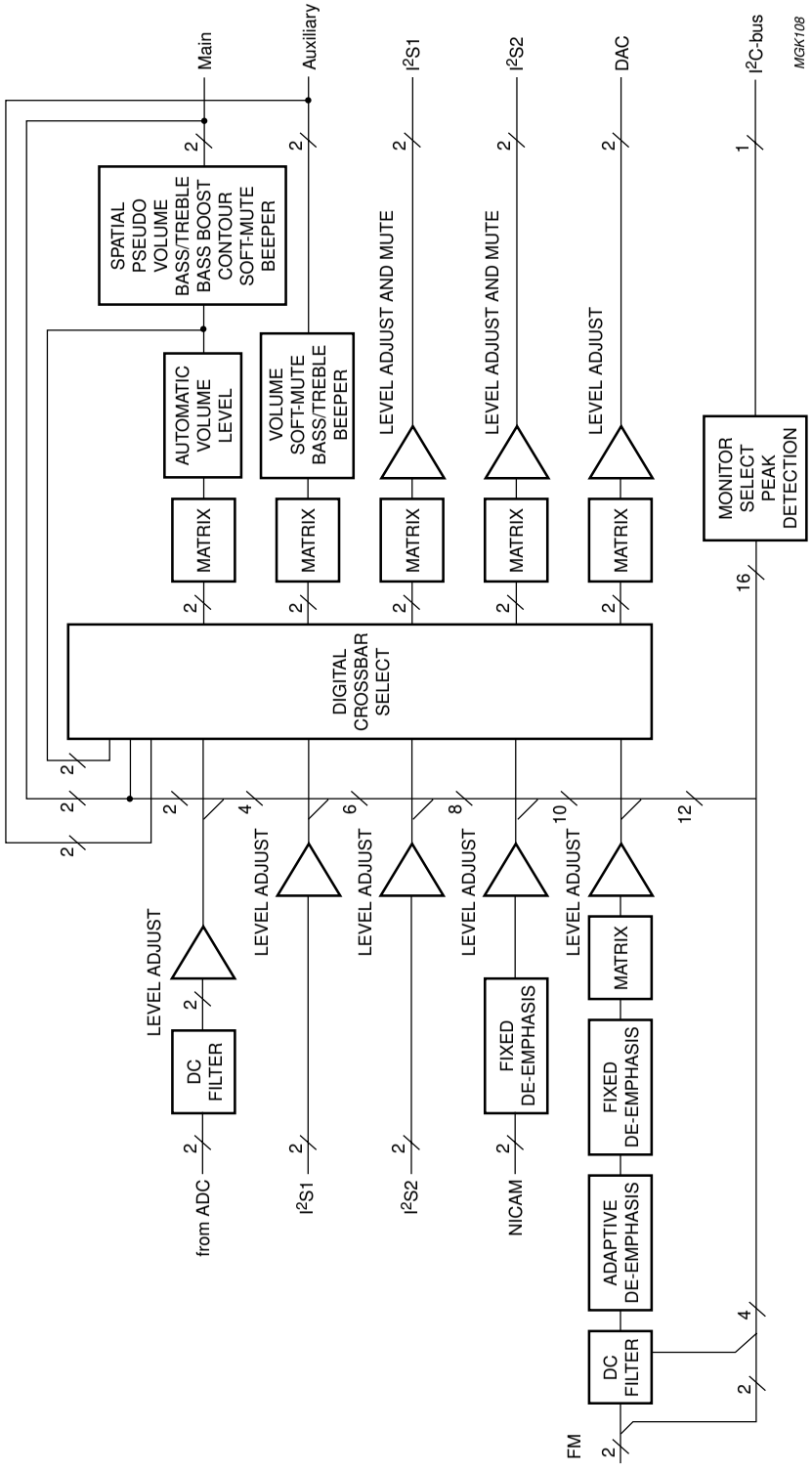


Fig.5 DSP data flow diagram.

6.2.1 LEVEL SCALING

All input channels to the digital crossbar switch (except for the loudspeaker feedback path) are equipped with a level adjust facility to change the signal level in a range from +15 to −15 dB (see Fig.5). It is recommended to scale all input channels to be 15 dB below full-scale (−15 dB full-scale) under nominal conditions.

6.2.2 NICAM PATH

The NICAM path has a switchable J17 de-emphasis.

6.2.3 FM (AM) PATH

A high-pass filter suppresses DC offsets from the FM demodulator due to carrier frequency offsets and supplies the monitor/peak function with DC values and an unfiltered signal, e.g. for the purpose of carrier detection.

The de-emphasis function offers fixed settings for the supported standards (50, 60 or 75 μs and J17).

An adaptive de-emphasis is available for Wegener-Panda 1 encoded programs.

A matrix performs the dematrixing of the A2 stereo, dual and mono signals.

6.2.4 NICAM AUTO-MUTE

If NICAM B/G, I or D/K is received, the auto-mute is enabled and the signal quality becomes poor, the digital crossbar switch switches automatically to FM and switches the matrix to channel 1. The automatic switching depends on the NICAM bit error rate.

The auto-mute function can be disabled via the I²C-bus.

For NICAM L applications, it is recommended to demodulate AM sound in the first sound IF and connect the audio signal to the mono input of the TDA9875A. By setting bit AMSEL (see Section 10.3.11), the auto-mute function will switch to the audio ADC instead of switching to the first sound carrier. The ADC source selector bits (see Section 10.3.20) should be set to mono input, where the AM sound signal should be connected.

6.2.5 MONITOR

This function provides data words from a number of locations in the signal processing paths to the I²C-bus interface (2 data bytes). Signal sources include the FM demodulator outputs, most inputs to the digital crossbar switch and the outputs of the ADC. Source selection and data read-out is performed via the I²C-bus. Optionally, the peak value can be measured instead of simply taking samples. The internally stored peak value is reset to zero when the data is read via the I²C-bus. The monitor function may be used, for example, for signal level measurements or carrier detection.

6.2.6 LOUDSPEAKER (MAIN) CHANNEL

The matrix provides the following functions: forced mono, stereo, channel swap, channel 1, channel 2 and spatial effects.

There are fixed coefficient sets for spatial settings of 30%, 40% and 52%.

The Automatic Volume Level (AVL) function provides a constant output level of −23 dB (full-scale) for input levels between 0 and −29 dB (full-scale). There are some fixed decay time constants to choose from, i.e. 2, 4 and 8 s.

Pseudo stereo is based on a phase shift in one channel via a second-order all-pass filter. There are fixed coefficient sets to provide 90 degrees phase shift at frequencies of 150, 200 and 300 Hz.

Volume is controlled individually for each channel ranging from +24 to −83 dB with 1 dB resolution. There is also a mute position. For the purpose of a simple control software in the microcontroller, the decimal number that is sent as an I²C-bus data byte for volume control is identical to the volume setting in dB (e.g. the I²C-bus data byte +10 sets the new volume value to +10 dB).

Balance can be realized by independent control of the left and right channel volume settings.

Contour is adjustable between 0 and +18 dB with 1 dB resolution. This function is linked to the volume setting by means of microcontroller software.

Bass is adjustable between +15 and −12 dB with 1 dB resolution and treble is adjustable between +12 and −12 dB with 1 dB resolution.

For the purpose of a simple control software in the microcontroller, the decimal number that is sent as an I²C-bus data byte for contour, bass or treble is identical to the new contour, bass or treble setting in dB (e.g. the I²C-bus data byte +8 sets the new value to +8 dB).

Extra bass boost is provided up to 20 dB with 2 dB resolution. The implemented coefficient set serves merely as an example on how to use this filter.

The beeper provides tones in a range from approximately 400 Hz to 30 kHz. The frequency can be selected via the I²C-bus. The beeper output signal is added to the loudspeaker and headphone channel signals.

The beeper volume is adjustable with respect to full-scale between 0 and −93 dB with 3 dB resolution. The beeper is not effected by mute.

Soft mute provides a mute ability in addition to volume control with a well defined time (32 ms) after which the soft mute is completed. A smooth fading is achieved by a cosine masking.

6.2.7 HEADPHONE (AUXILIARY) CHANNEL

The matrix provides the following functions: forced mono, stereo, channel swap, channel 1 and channel 2 (or C and S in Dolby Surround Pro Logic mode).

Volume is controlled individually for each channel in a range from +24 to −83 dB with 1 dB resolution. There is also a mute position. For the purpose of a simple control software in the microcontroller, the decimal number that is sent as an I²C-bus data byte for volume control is identical to the volume setting in dB (e.g. the I²C-bus data byte +10 sets the new volume value to +10 dB).

Balance can be realized by independent control of the left and right channel volume settings.

Bass is adjustable between +15 and −12 dB with 1 dB resolution and treble is adjustable between +12 and −12 dB with 1 dB resolution. For the purpose of a simple control software in the microcontroller, the decimal number that is sent as an I²C-bus data byte for bass or treble is identical to the new bass or treble setting in dB (e.g. the I²C-bus data byte +8 sets the new value to +8 dB).

The beeper provides tones in a range from approximately 400 Hz to 30 kHz. The frequency can be selected via the I²C-bus. The beeper output signal is added to the loudspeaker and headphone channel signals. The beeper volume is adjustable with respect to full-scale between 0 and −93 dB with 3 dB resolution. The beeper is not effected by mute.

Soft mute provides a mute ability in addition to volume control with a well defined time (32 ms) after which the soft mute is completed. A smooth fading is achieved by a cosine masking.

6.2.8 FEATURE INTERFACE

The feature interface comprises two I²S-bus input/output ports and a system clock output. Each I²S-bus port is equipped with level adjust facilities that can change the signal level in a range from +15 to −15 dB with 1 dB resolution. Outputs can be disabled to improve EMC performance.

The I²S-bus output matrix provides the following functions: forced mono, stereo, channel swap, channel 1 and channel 2.

One example of how the feature interface can be used in a TV set is to connect an external Dolby Surround Pro Logic DSP, such as the SAA7710, to the I²S-bus ports. Outputs must be enabled and a suitable master clock signal for the DSP can be taken from pin SYSCLK. A stereo signal from any source will be output on one of the I²S-bus serial data outputs and the four processed signal channels will be entered at both I²S-bus serial data inputs. Left and right could then be output to the power amplifiers via the Main channel, centre and surround via the Auxiliary channel.

6.2.9 CHANNEL FROM THE AUDIO ADC

The signal level at the output of the ADC can be adjusted in a range from +15 to −15 dB with 1 dB resolution. The audio ADC itself is scaled to a gain of −6 dB.

6.2.10 CHANNEL TO THE ANALOG CROSSBAR PATH

Level adjust with control positions 0, +3, +6 and +9 dB.

6.2.11 DIGITAL CROSSBAR SWITCH

Input channels to the crossbar switch are from the audio ADC, I²S1, I²S2, FM path, NICAM path and from the loudspeaker channel path after matrix and AVL (see Fig.8).

Output channels comprise loudspeaker, headphone, I²S1, I²S2 and audio DACs for line output and SCART. I²S1 and I²S2 outputs also provide digital outputs from the loudspeaker and headphone channels, but without the beeper signals.

6.2.12 SIGNAL GAIN

There are a number of functions that can provide signal gain, e.g. volume, bass and treble control. Great care has to be taken when using gain with large input signals in order not to exceed the maximum possible signal swing, which would cause severe signal distortion. The nominal signal level of the various signal sources to the digital crossbar switch should be 15 dB below digital full-scale (−15 dB full-scale). This means that a volume setting of, say, +15 dB would just produce a full-scale output signal and not cause clipping, if the signal level is nominal.

Sending illegal data patterns via the I²C-bus will not cause any changes of the current setting for the volume, bass, treble, bass boost and level adjust functions.

6.2.13 EXPERT MODE

The TDA9875A provides a special expert mode that gives direct write access to the internal Coefficient RAM (CRAM) of the DSP. It can be used to create user-defined characteristics, such as a tone control with different corner frequencies or special boost/cut characteristics to correct the low-frequency loudspeaker and/or cabinet frequency responses by means of the bass boost filter. However, this mode must be used with great care.

More information on the functions of this device, such as the number of coefficients per function, their default values, memory addresses, etc., can be made available on request.

6.2.14 DSP FUNCTIONS

Table 5 Overview of DSP functions

FUNCTION	EXPERT MODE	PARAMETER	VALUE	UNIT
Bass control for loudspeaker and headphone output	yes	control range	−12 to +15	dB
		resolution	1	dB
		resolution at frequency	40	Hz
Treble control for loudspeaker and headphone output	yes	control range	−12 to +12	dB
		resolution	1	dB
		resolution at frequency	14	kHz
Contour for loudspeaker output	yes	control range	0 to +18	dB
		resolution	1	dB
		resolution at frequency	40	Hz
Bass boost for loudspeaker output	yes	control range	0 to +20	dB
		resolution	2	dB
		resolution at frequency	20	Hz
		corner frequency	350	Hz
Volume control for each separate channel in loudspeaker and headphone output	no	control range	−83 to +24	dB
		resolution	1	dB
		mute position at step	1010 1100	
Soft mute for loudspeaker and headphone output	no	processing time	32	ms
Spatial effects	yes	anti-phase crosstalk positions	30, 40 and 52	%
Pseudo stereo	yes	90 degrees phase shift at frequency	150, 200 and 300	Hz
Beeper additional to the signal in the loudspeaker and headphone channel	yes	beep frequencies	see Section 10.3.38	
		control range	0 to −93	dB
		resolution	3	dB
		mute position at step	0010 0000	
Automatic Volume Level (AVL)	yes	step width	quasi continuously	
		AVL output level for an input level between 0 and −29 dB (full-scale)	−23	dB
		attack time	10	ms
		decay time constant	2, 4 and 8	s

FUNCTION	EXPERT MODE	PARAMETER	VALUE	UNIT
General	no	−3 dB lower corner frequency of DSP	10	Hz
		−1 dB bandwidth of DSP	14.5	kHz
Level adjust I^2S1 and I^2S2 inputs	yes	control range	−15 to +15	dB
		resolution	1	dB
Level adjust I^2S1 and I^2S2 outputs	yes	control range	−15 to +15	dB
		resolution	1	dB
		mute position at step	0001 0000	
Level adjust analog crossbar path	no	control positions	0, 3, 6 and 9	dB
Level adjust audio ADC outputs	yes	control range	+15 to −15	dB
		resolution	1	dB
Level adjust NICAM path	yes	control range	+15 to −15	dB
		resolution	1	dB
Level adjust FM path	yes	control range	+15 to −15	dB
		resolution	1	dB

6.3 Analog audio section

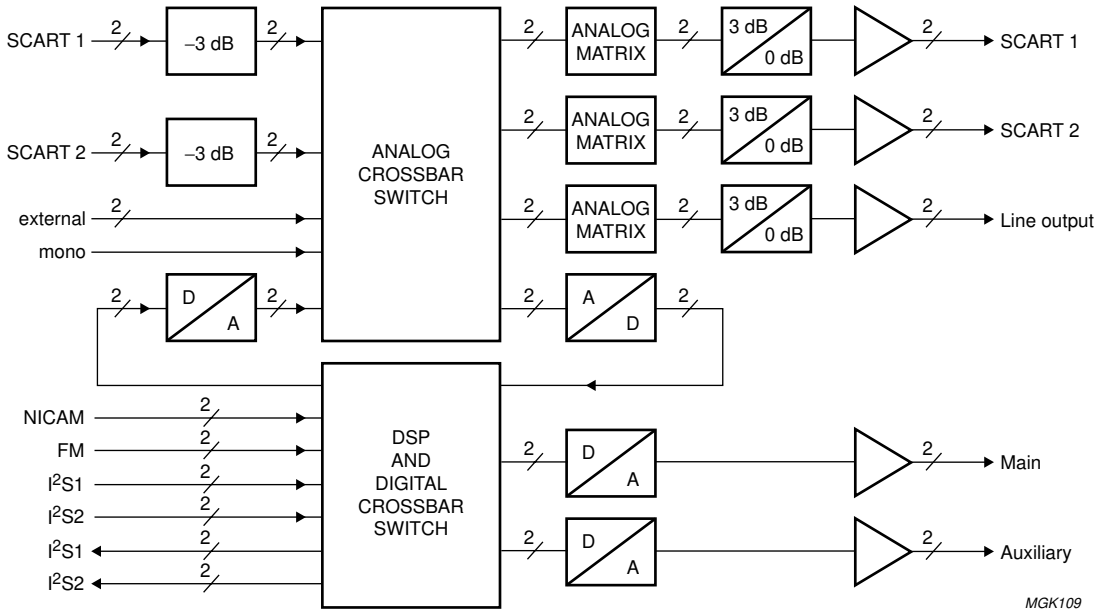


Fig.6 Block diagram for the audio section.

6.3.1 Analog crossbar switch and analog matrix

There are a number of analog input and output ports with the TDA9875A (see Figs 6 and 8). Analog source selector switches are employed to provide the desired analog signal routing capability. The analog signal routing is performed by the analog crossbar switch section. A dual audio ADC provides the connection to the DSP section and a dual audio DAC provides the connection from the DSP section to the analog crossbar switch. The digital signal routing is performed by a digital crossbar switch.

The basic signal routing philosophy of the TDA9875A is that each switch handles two signal channels at the same time, e.g. left and right, language A and B, directly at the source.

Each source selector switch is followed by an analog matrix to perform further selection tasks, such as putting a signal from one input channel, say language A, to both output channels or for swapping left and right channels (see Fig.7).

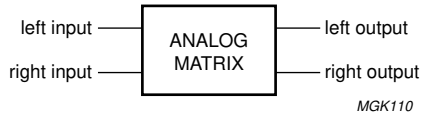


Fig.7 Analog matrix.

The analog matrix provides the functions given in Table 6.

Table 6 Analog matrix functions

MODE	MATRIX OUTPUT	
	LEFT OUTPUT	RIGHT OUTPUT
1	left input	right input
2	right input	left input
3	left input	left input
4	right input	right input

All switches and matrices are controlled via the I^2C-bus.

6.3.2 SCART inputs

The SCART specification allows for a signal level of up to 2 V (RMS). Because of signal handling limitations, due to the 5 V supply voltage of the TDA9875A, it is necessary to have fixed 3 dB attenuators at the SCART inputs to obtain a 2 V input. This results in a −3 dB SCART-to-SCART copy gain. If 0 dB copy gain is preferred (with a maximum input of 1.4 V), there are 0/3 dB amplifiers at the outputs of SCART 1 and SCART 2 and at the line output.

The input attenuator is realized by an external series resistor in combination with the input impedance, both of which form a voltage divider. With this voltage divider the maximum SCART signal level of 2 V (RMS) is scaled down to 1.4 V (RMS) at the input pin.

6.3.3 External and mono inputs

The 3 dB input attenuators are not required for the external and mono inputs, because those signal levels are under control of the TV designer. The maximum allowed input level is 1.4 V (RMS). By adding external series resistors, the external inputs can be used as an additional SCART input.

6.3.4 SCART outputs

The SCART outputs employ amplifiers with two gain settings. The gain can be set to 3 or 0 dB via the I^2C-bus. The 3 dB position is needed to compensate for the 3 dB attenuation at the SCART inputs should SCART-to-SCART copies with 0 dB gain be preferred [under the condition of 1.4 V (RMS) maximum input level]. The 0 dB position is needed, for example, for an external-to-SCART copy with 0 dB gain.

6.3.5 LINE OUTPUT

The line output can provide an unprocessed copy of the audio signal in the loudspeaker channels. This can be either an external signal that comes from the dual audio ADC, or a signal from an internal digital audio source that comes from the dual audio DAC. The line output employs amplifiers with two gain settings. The 3 dB position is needed to compensate for the attenuation at the SCART inputs, while the 0 dB position is needed, for example, for non-attenuated external or internal digital signals (see Section 6.3.4).

6.3.6 LOUDSPEAKER (MAIN) AND HEADPHONE (AUXILIARY) OUTPUTS

Signals from any audio source can be applied to the loudspeaker and to the headphone output channels via the digital crossbar switch and the DSP.

6.3.7 DUAL AUDIO DAC

The TDA9875A contains three dual audio DACs, one for the connection from the DSP to the analog crossbar switch section and two for the loudspeaker and headphone outputs. Each of the three dual low-noise high-dynamic range DACs consists of two 15-bit DACs with current outputs, followed by a buffer operational amplifier. The audio DACs operate with four-fold oversampling and noise shaping.

6.3.8 DUAL AUDIO ADC

There is one dual audio ADC in the TDA9875A for the connection of the analog crossbar switch section to the DSP. The dual audio ADC consists of two bitstream third-order sigma-delta audio ADCs and a high-order decimation filter.

6.3.9 STANDBY MODE

The standby mode, selected by setting bit STDBY to logic 1 (see Section 10.3.2) disables most functions and reduces power dissipation. The analog crossbar switch and the SCART section remain operational and can be controlled by the I^2C-bus to support copying of analog signals from SCART-to-SCART.

Unused internal registers may lose their information in the standby mode. Therefore, the device needs to be initialized on returning to the normal operating mode. This can be accomplished in the same way as after a Power-on reset.

6.3.10 SUPPLY GROUND

The different supply grounds V_{SS} are internally connected via the substrate. It is recommended to connect all ground pins by means of a copper plane close to the pins.

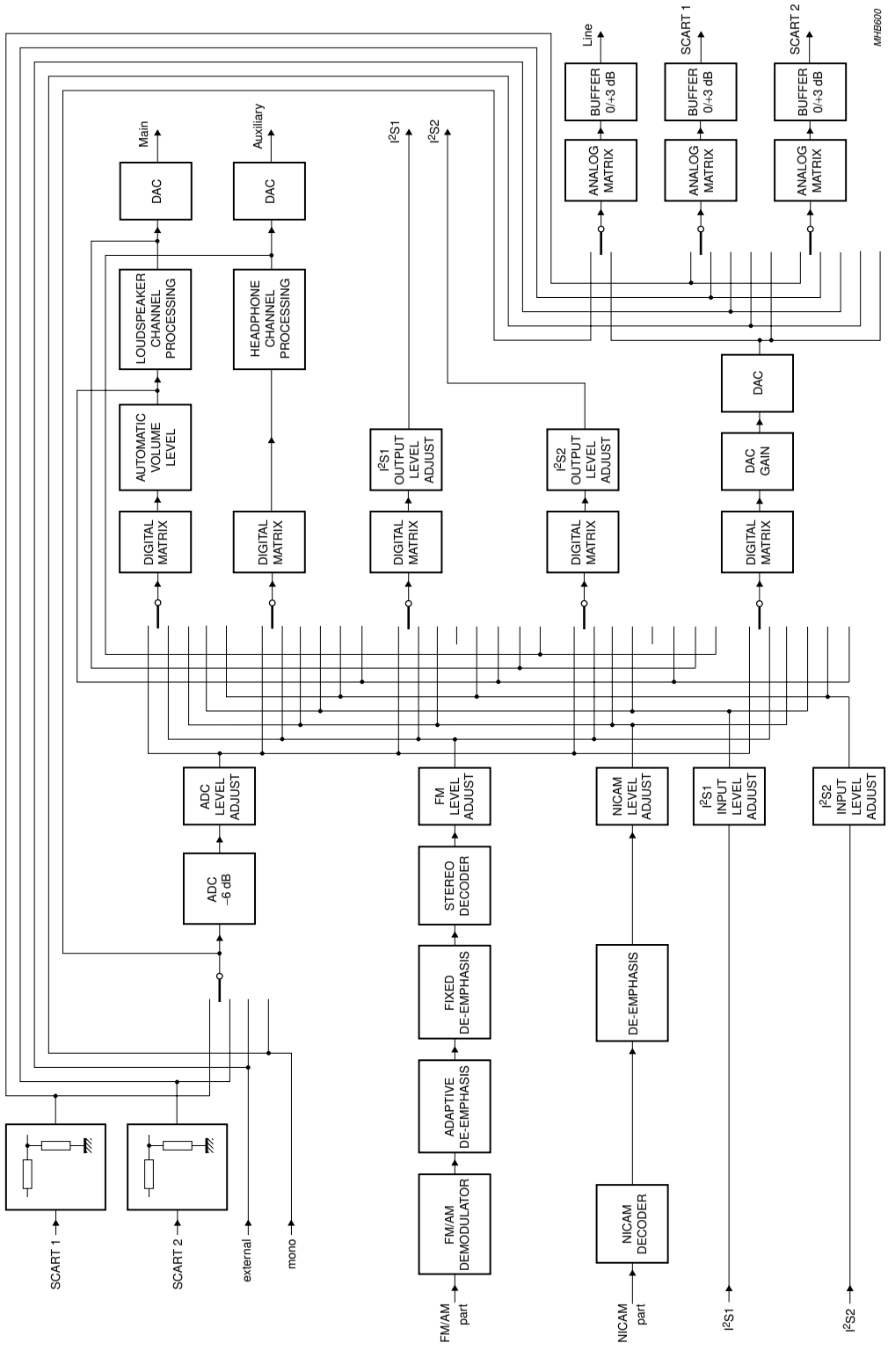


Fig.8 Audio signal flow diagram.

7 LIMITING VALUES

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

SYMBOL	PARAMETER	CONDITIONS	MIN.	MAX.	UNIT
V_{DD}	DC supply voltage		−0.5	+6.0	V
ΔV_{DD}	voltage differences between two V_{DD} pins		−	550	mV
V_n	voltage on any other pin		−0.5	V_{DD} + 0.5	V
I_{DDD}, I_{SSD}	DC current per digital supply pin		−	±180	mA
$I_{lu(prot)}$	latch-up protection current		100	−	mA
P_{tot}	total power dissipation		−	1.0	W
T_{stg}	storage temperature		−55	+125	°C
T_{amb}	ambient temperature		−20	+70	°C
V_{es}	electrostatic handling voltage	note 1	−2000	+2000	V
		note 2	−200	+200	V

Notes

1. Human body model: C = 100 pF; R = 1.5 kΩ.
2. Machine model: C = 200 pF; L = 0.75 μH; R = 0 Ω.

8 THERMAL CHARACTERISTICS

SYMBOL	PARAMETER	CONDITIONS	VALUE	UNIT
$R_{th(j-a)}$	thermal resistance from junction to ambient	in free air		
	TDA9875A (SDIP64)		40	K/W
	TDA9875AH (QFP64)		50	K/W

9 CHARACTERISTICS

$V_{SIF(p-p)}$ = 300 mV; AGCOFF = 0; AGCSLOW = 0; AGCLEV = 0; level and gain settings in accordance with note 1; V_{DD} = 5 V; T_{amb} = 25 °C; settings in accordance with B/G standard; FM deviation ±50 kHz; f_{mod} = 1 kHz; FM sound parameters in accordance with system A2; NICAM in accordance with *"EBU specification"*; 1 kΩ measurement source resistance for AF inputs; with external components of Fig.10; unless otherwise specified.

SYMBOL	PARAMETER	CONDITIONS	MIN.	TYP.	MAX.	UNIT
Supplies						
V_{DDD1}	digital supply voltage 1		4.75	5.0	5.5	V
V_{SSD1}	digital supply ground 1	note 2	–	0.0	–	V
I_{DDD1}	digital supply current 1	V_{DDD1} = 5.0 V	58	73	88	mA
V_{DDD2}	digital supply voltage 2		4.75	5.0	5.5	V
V_{SSD2}	digital supply ground 2	note 2	–	0.0	–	V
I_{DDD2}	digital supply current 2	V_{DDD2} = 5.0 V; system clock output disabled	0.1	0.4	2	mA
V_{SSD3}	digital supply ground 3	note 2	–	0.0	–	V
V_{SSD4}	digital supply ground 4	note 2	–	0.0	–	V
V_{DDA}	analog supply voltage		4.75	5.0	5.5	V
I_{DDA}	analog supply current for DAC part	V_{DDA} = 5.0 V; digital silence	44	56	68	mA
V_{SSA1}	analog ground for analog front-end	note 2	–	0.0	–	V
V_{SSA2}	analog ground for audio ADC part	note 2	–	0.0	–	V
V_{SSA3}	analog ground for audio DAC part	note 2	–	0.0	–	V
V_{SSA4}	analog ground for SCART		–	0.0	–	V
Demodulator supply decoupling and references						
V_{DEC1}	analog supply decoupling voltage for demodulator part		3.0	3.3	3.6	V
V_{ref1}	analog reference voltage for demodulator part		–	2	–	V
$I_{ref1(sink)}$	sink current at pin V_{ref1}		–	200	–	μA
Audio supply decoupling and references						
V_{DEC2}	analog supply decoupling voltage for audio ADC part		3.0	3.3	3.6	V
V_{ref2}	reference voltage ratio for audio ADCs	referenced to V_{DEC2} and V_{SSA2}	–	50	–	%
$Z_{Vref2-VDEC2}$	impedance pins V_{ref2} to V_{DEC2}		–	20	–	kΩ
$Z_{Vref2-VSSA2}$	impedance pins V_{ref2} to V_{SSA2}		–	20	–	kΩ
V_{ref3}	reference voltage ratio for audio DAC and operational amplifier	referenced to V_{DDA} and V_{SSA3}	–	50	–	%
$Z_{Vref3-VDDA}$	impedance pins V_{ref3} to V_{DDA}		–	20	–	kΩ
$Z_{Vref3-VSSA3}$	impedance pins V_{ref3} to V_{SSA3}		–	20	–	kΩ

SYMBOL	PARAMETER	CONDITIONS	MIN.	TYP.	MAX.	UNIT
Power fail detector						
$V_{th(pf)}$	power fail threshold voltage		–	3.9	–	V
Digital inputs and outputs						
INPUTS						
CMOS level input, pull-down (pins TEST1 and TEST2)						
V_{IL}	LOW-level input voltage		–	–	$0.3V_{DDD}$	V
V_{IH}	HIGH-level input voltage		$0.7V_{DDD}$	–	–	V
C_i	input capacitance		–	–	10	pF
Z_i	input impedance		–	50	–	kΩ
CMOS level input, hysteresis, pull-up (pin CRESET)						
V_{IL}	LOW-level input voltage		–	–	$0.3V_{DDD}$	V
V_{IH}	HIGH-level input voltage		$0.7V_{DDD}$	–	–	V
V_{hys}	hysteresis voltage		–	1.3	–	V
C_i	input capacitance		–	–	10	pF
Z_i	input impedance		30	50	–	kΩ
INPUTS/OUTPUTS						
I^2C-bus level input with Schmitt trigger, open-drain output stage, 400 kHz I^2C-bus operation (pins SCL and SDA)						
V_{IL}	LOW-level input voltage		–	–	$0.3V_{DDD}$	V
V_{IH}	HIGH-level input voltage		$0.7V_{DDD}$	–	–	V
V_{hys}	hysteresis voltage		–	$0.05V_{DDD}$	–	V
I_{LI}	input leakage current		–	–	±10	µA
C_i	input capacitance		–	–	10	pF
V_{OL}	LOW-level output voltage		–	–	0.6	V
C_L	load capacitance		–	–	400	pF
TTL/CMOS level, 4 mA 3-state output stage, pull-up (pins PCLK, NICAM, ADDR1, ADDR2, P1, P2, SCK, WS, SDO1, SDO2, SDI1 and SDI2)						
V_{IL}	LOW-level input voltage		–	–	0.8	V
V_{IH}	HIGH-level input voltage		2.0	–	–	V
C_i	input capacitance		–	–	10	pF
V_{OL}	LOW-level output voltage		–	–	0.4	V
V_{OH}	HIGH-level output voltage		2.4	–	–	V
C_L	load capacitance		–	–	100	pF
Z_i	input impedance		–	50	–	kΩ
OUTPUTS						
CMOS level output, 4 mA 3-state output stage, slew rate controlled (pin SYSCLK)						
V_{OL}	LOW-level output voltage		–	–	$0.3V_{DDD}$	V
V_{OH}	HIGH-level output voltage		$0.7V_{DDD}$	–	–	V
C_L	load capacitance		–	–	100	pF
I_{LIZ}	3-state leakage current	V_i = 0 to V_{DDD}	–	–	±10	µA

SYMBOL	PARAMETER	CONDITIONS	MIN.	TYP.	MAX.	UNIT
SIF1 and SIF2 analog inputs						
$V_{SIF(max)(p-p)}$	maximum composite SIF input voltage for clipping (peak-to-peak value)	SIF input level adjust 0 dB	–	941	–	mV
		SIF input level adjust −10 dB	–	2976	–	mV
$V_{SIF(min)(p-p)}$	minimum composite SIF input voltage for lower limit of AGC (peak-to-peak value)	SIF input level adjust 0 dB	–	59	–	mV
		SIF input level adjust −10 dB	–	188	–	mV
AGC	AGC range		–	24	–	dB
f_i	input frequency		4	–	9.2	MHz
R_i	input resistance	AGCLEV = 0	10	–	–	kΩ
C_i	input capacitance		–	7.5	11	pF
Δf_{FM}	FM deviation	B/G standard; THD < 1%	±100	–	–	kHz
$\Delta f_{FM(FS)}$	FM deviation full-scale level	terrestrial FM; level adjust 0 dB	±150	–	–	kHz
C/N_{FM}	FM carrier-to-noise ratio	N_{FM} bandwidth = 6 MHz; white noise for S/N = 40 dB; *"CCIR468"*; quasi peak	–	77	–	$\frac{dB}{Hz}$
C/N_N	NICAM carrier-to-noise ratio	N_N bandwidth = 6 MHz; bit error rate = 10^{-3}; white noise	–	66	–	$\frac{dB}{Hz}$
α_{ct}	crosstalk attenuation SIF1 to SIF2	f_i = 4 to 9.2 MHz; note 3	50	–	–	dB
Demodulator performance						
THD + N	total harmonic distortion plus noise	from FM source to any output; V_o = 1 V (RMS) with low-pass filter	–	0.3	0.5	%
		from NICAM source to any output; V_o = 1 V (RMS) with low-pass filter	–	0.1	0.3	%
S/N	signal-to-noise ratio	SC1 from FM source to any output; V_o = 1 V (RMS); *"CCIR468"*; quasi peak	64	70	–	dB
		SC2 from FM source to any output; V_o = 1 V (RMS); *"CCIR468"*; quasi peak	60	66	–	dB
		NICAM source; V_o = 1 V (RMS); note 4	–	–	–	
B_{-3dB}	−3 dB bandwidth	from FM source to any output	14.5	15	–	kHz
		from NICAM source to any output	14.5	15	–	kHz
f_{res}	frequency response 20 Hz to 14 kHz	from FM or NICAM to any output; f_{ref} = 1 kHz; inclusive pre-emphasis and de-emphasis	–	±2	–	dB

SYMBOL	PARAMETER	CONDITIONS	MIN.	TYP.	MAX.	UNIT
$\alpha_{cs(dual)}$	dual signal channel separation	note 5	65	70	–	dB
$\alpha_{cs(stereo)}$	stereo channel separation	note 6	40	45	–	dB
α_{AM}	AM suppression for FM	AM: 1 kHz, 30% modulation; reference: 1 kHz, 50 kHz deviation	50	–	–	dB
S/N_{AM}	AM demodulation	SIF level 100 mV (RMS); 54% AM; 1 kHz AF; *"CCIR468"*; quasi peak	36	45	–	dB
IDENTIFICATION FOR FM SYSTEMS						
mod_p	pilot modulation for identification		25	50	75	%
C/N_p	pilot sideband carrier-to-noise ratio for identification start		–	27	–	$\frac{dB}{Hz}$
f_{ident}	identification window	B/G stereo				
		slow mode	116.85	–	118.12	Hz
		medium mode	116.11	–	118.89	Hz
		fast mode	114.65	–	120.46	Hz
		B/G dual				
		slow mode	273.44	–	274.81	Hz
		medium mode	272.07	–	276.20	Hz
		fast mode	270.73	–	277.60	Hz
$t_{ident(on)}$	total identification time ON	slow mode	–	–	2	s
		medium mode	–	–	1	s
		fast mode	–	–	0.5	s
$t_{ident(off)}$	total identification time OFF	slow mode	–	–	2	s
		medium mode	–	–	1	s
		fast mode	–	–	0.5	s
Analog audio inputs						
MONO INPUT AND EXTERNAL INPUT						
$V_{i(nom)(rms)}$	nominal level input voltage (RMS value)		–	500	–	mV
$V_{i(clip)(rms)}$	clipping level input voltage (RMS value)	THD < 3%; note 7	1250	1400	–	mV
R_i	input resistance	note 7	28	35	42	kΩ
SCART INPUTS						
$V_{i(nom)(rms)}$	nominal level input voltage at input pin (RMS value)	−3 dB divider with external 15 kΩ resistor; note 8	–	350	–	mV
$V_{i(clip)(rms)}$	clipping level input voltage at input pin (RMS value)	−3 dB divider with external 15 kΩ resistor; THD < 3%; notes 7 and 8	1250	1400	–	mV
R_i	input resistance	note 7	28	35	42	kΩ