DATA SHEET



SAA7715H
Digital Signal Processor

Preliminary specification
File under Integrated Circuits, IC01

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CONTENTS

1 FEATURES
1.1 Hardware
1.2 Possible firmware
2 APPLICATIONS
3 GENERAL DESCRIPTION
4 QUICK REFERENCE DATA
5 ORDERING INFORMATION
6 BLOCK DIAGRAM
7 PINNING
8 FUNCTIONAL DESCRIPTION
8.1 PLL division factors for different clock inputs
8.2 The word select PLL
8.3 The Filter Stream DAC (FSDAC)
8.3.1 Interpolation filter
8.3.2 Noise shaper
8.3.3 Function of pin POM
8.3.4 Power off plop suppression
8.3.5 Pin VREFDA for internal reference
8.3.6 Supply of the analog outputs
8.4 External control pins
8.5 Digital serial inputs/outputs and SPDIF inputs
8.5.1 Digital serial inputs/outputs
8.5.2 SPDIF inputs
8.6 I²C-bus interface (pins SCL and SDA)
8.7 Reset
8.8 Power-down mode
8.9 Power supply connection and EMC
8.10 Test mode connections (pins TSCAN, RTCB and SHTCB)
9 I²C-BUS PROTOCOL
9.1 Addressing
9.2 Slave address (pin A0)
9.3 Write cycles
9.4 Read cycles
9.5 Program RAM
9.6 Data word alignment
9.7 I²C-bus memory map specification
9.8 I²C-bus memory map definition
9.9 Table definitions
10 SOFTWARE IN ROM DESCRIPTION
10.1 Audio dynamics compressor
10.1.1 Theory of operation
10.2 Audio enhancer
10.2.1 Theory of operation
10.3 Equalizer
10.3.1 General description
10.3.2 Overview
10.3.3 Controls
10.3.4 Centre frequency
10.3.5 Gain
10.3.6 Q
10.3.7 Hints and tips
10.4 Stereo spatializer
10.4.1 Overview
10.4.2 Controls
10.4.3 Mix
10.4.4 Hints and tips
11 LIMITING VALUES
12 THERMAL CHARACTERISTICS
13 CHARACTERISTICS
14 I²S-BUS TIMING
15 I²C-BUS TIMING
16 APPLICATION DIAGRAM
17 PACKAGE OUTLINE
18 SOLDERING
18.1 Introduction to soldering surface mount packages
18.2 Reflow soldering
18.3 Wave soldering
18.4 Manual soldering
18.5 Suitability of surface mount IC packages for wave and reflow soldering methods
19 DATA SHEET STATUS
20 DEFINITIONS
21 DISCLAIMERS
22 PURCHASE OF PHILIPS I²C COMPONENTS

1 FEATURES

1.1 Hardware

- 24-bit Philips 70 MIPS DSP core (24-bit data path and 12/24-bit coefficient path)
- 1.5 kbyte of downloadable DSP program memory (PRAM)
- 2 kbyte of DSP program memory (PROM)
- 2.5 kbyte of re-programmable DSP data memory (XRAM)
- 512 byte of re-programmable DSP coefficient memory (YRAM)
- Four stereo digital serial inputs (8 channels) with common BCK and WS. To these inputs the I^2S-bus format or LSB-justified formats can be applied
- One stereo bitstream DAC (2 channels) with 64 fold oversampling and noise shaping
- Selectable clock output (pin SYSCLK) for external slave devices ($512f_s$ to $128f_s$)
- Four stereo digital serial outputs (8 channels) with selectable I^2S-bus or LSB-justified format
- Two SPDIF inputs combined with digital serial input
- On-board WS_PLL generates clock for on-board DAC and output pin SYSCLK
- I^2C-bus controlled (including fast mode)
- Programmable Phase-Locked Loop (PLL) derives the clock for the DSP from the CLK_IN input
- –40 to +85 °C operating temperature range
- supply voltage only 3.3 V
- All digital inputs are tolerant for 5 V input levels
- Power-down mode for low current consumption in standby mode
- Optimized pinning for applications with other Philips DACs (such as UDA1334, UDA1355 and UDA1328).

1.2 Possible firmware

- Dolby®(1) Pro Logic decoding
- Smoothed volume control (without zipper noise)
- Automatic Volume Levelling (AVL)
- Dynamic bass enhancement
- Ultra bass
- Incredible surround
- Incredible mono (Imono)
- DPL virtualiser
- Dolby digital virtualiser (DVD post-processing)
- Dynamic compressor
- Spectral enhancer
- Equalizer with peaking/shelving filters
- DC filters
- Bass/treble control
- Dynamic loudness
- Tone/noise generator
- Graphical spectrum analyser
- Configurable Delay Unit (DLU)
- Sound steering/elevation for CAR applications
- Sample Rate Conversion (SRC).

2 APPLICATIONS

- As co-processor for a car radio DSP in a car radio application for additional acoustic enhancements (sound steering/sound elevation/signal processing)
- Multichannel audio: in DVD and Home theatre applications as post-processing device like signal virtualisation (virtual 3D surround) and acoustic enhancement, tone control, volume control and equalizers
- Multichannel decoding: Dolby Pro Logic and virtual 3D surround
- PC/USB audio applications: stereo widening (Incredible surround), sound steering, sound positioning and speaker equalization.

(1) **Dolby** — Available only to licensees of Dolby Laboratories Licensing Corporation, San Francisco, CA94111, USA, from whom licensing and application information must be obtained. Dolby is a registered trade-mark of Dolby Laboratories Licensing Corporation.

3 GENERAL DESCRIPTION

The SAA7715 is a cost effective and powerful high performance 24-bit programmable DSP for a variety of digital audio applications. This DSP device integrates a 24-bit DSP core with programmable memories (program RAM/ROM, data and coefficient RAM), 4 digital serial inputs, 4 digital serial outputs, 2 separate SPDIF receivers, a stereo FSDAC, a standard Philips I^2C-bus interface, a phase-locked loop for the DSP clock generation and a second phase-locked loop for system clock generation (internal and external DAC clocks).

The SAA7715 can be configured for various audio applications by downloading the dedicated DSP program code into the DSP program RAM or using the ROM or a combination of both. During the 'Power-down mode' the contents of the memories and all other settings will keep their values. The SAA7715 can be initialized using the I^2C-bus interface.

Several system application examples, based on this existing SAA7715, are available for a wide range of audio applications (e.g car radio DSP, DVD post-processing, Dolby Pro Logic, PC/USB audio and more) which can be used as a reference design for customers.

4 QUICK REFERENCE DATA

SYMBOL	PARAMETER	CONDITIONS	MIN.	TYP.	MAX.	UNIT
V_{DD}	operating supply voltage	all pins V_{DD} with respect to pins V_{SS}	3.15	3.3	3.45	V
I_{DDD}	supply current of the digital part	high activity of the DSP at DSPFREQ frequency	–	95	–	mA
I_{DDA}	supply current of the analog part	zero input and output signal	–	20	–	mA
P_{tot}	total power dissipation	high activity of the DSP at DSPFREQ frequency	–	380	–	mW
$I_{POWERDOWN}$	DC supply current of the total chip in Power-down mode	pin POWERDOWN enabled	–	400	–	µA
f_s	sample frequency	at IIS_WS1, SPDIF1 or SPDIF2 input	32	44.1	96	kHz
(THD + N)/S_{DAC}	total harmonic distortion-plus-noise to signal ratio of DAC	at 0 dB	–	–85	–	dB(A)
		at –60 dB	–	–37	–	dB(A)
S/N_{DAC}	signal-to-noise ratio of DAC	code = 0	–	100	–	dB(A)
CLK_IN	clock input	DIV_CLK_IN = LOW	8.192	11.2896	12.288	MHz
		DIV_CLK_IN = HIGH	16.384	–	24.576	MHz
DSPFREQ	maximum DSP clock		–	–	70	MHz

5 ORDERING INFORMATION

TYPE NUMBER	PACKAGE		
	NAME	DESCRIPTION	VERSION
SAA7715H	QFP44	plastic quad flat package; 44 leads (lead length 1.3 mm); body 10 × 10 × 1.75 mm	SOT307-2

6 BLOCK DIAGRAM

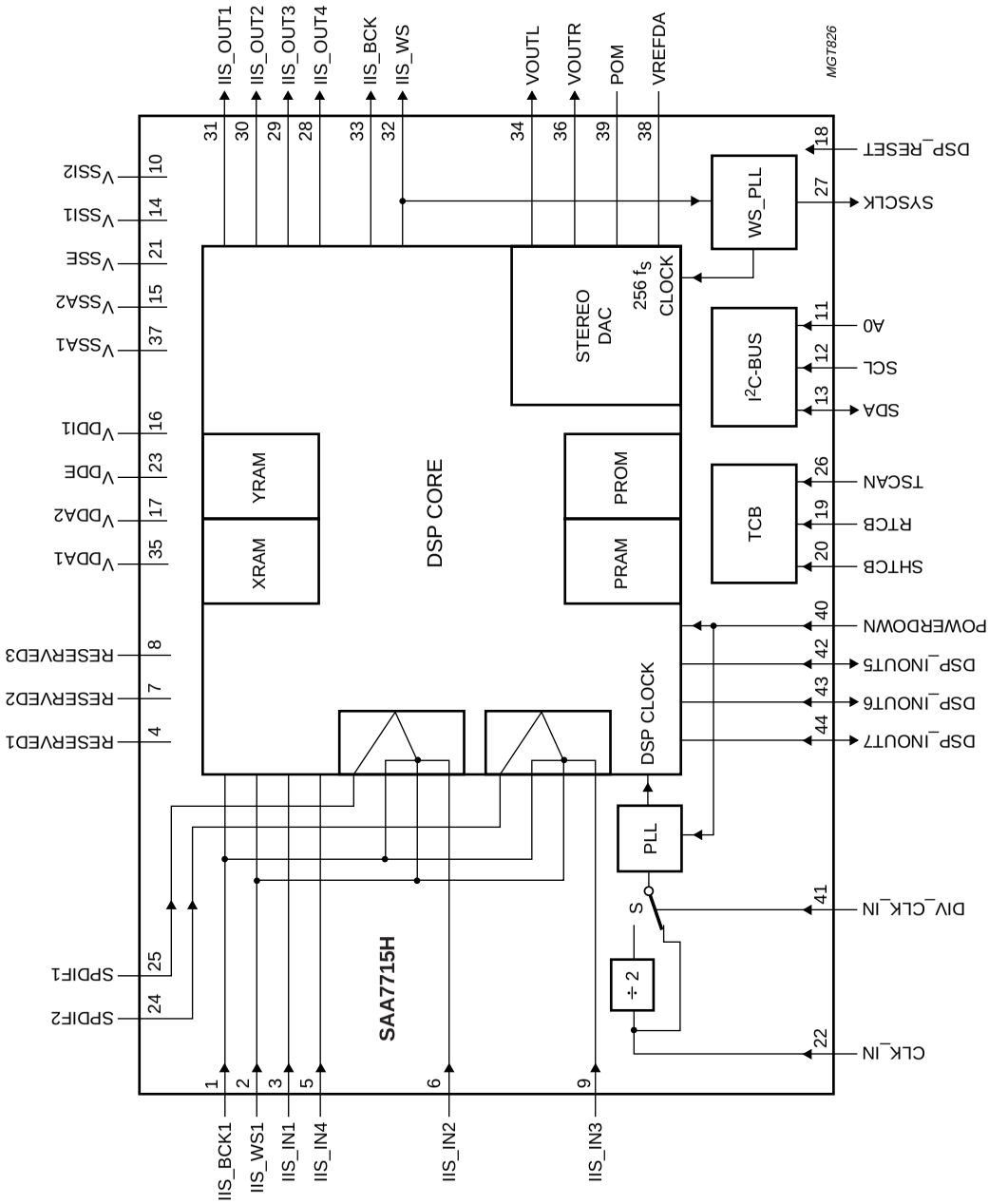


Fig.1 Block diagram.

7 PINNING

SYMBOL	PIN	PIN TYPE	DESCRIPTION
IIS_BCK1	1	ipthdt5v	bit clock signal belonging to data of digital serial inputs 1 to 4
IIS_WS1	2	ipthdt5v	word select signal belonging to data of digital serial inputs 1 to 4
IIS_IN1	3	ipthdt5v	data pin of digital serial input 1
RESERVED1	4	ipthdt5v	not to be connected externally
IIS_IN4	5	ipthdt5v	data pin of digital serial input 4
IIS_IN2	6	ipthdt5v	data pin of digital serial input 2
RESERVED2	7	ipthdt5v	not to be connected externally
RESERVED3	8	ipthdt5v	not to be connected externally
IIS_IN3	9	ipthdt5v	data pin of digital serial input 3
V_{SSI2}	10	vssi	ground supply (core only) (bond out to 2 pads)
A0	11	ipthdt5v	slave sub-address I^2C-bus selection/serial data input test control block
SCL	12	iptht5v	clock input of I^2C-bus
SDA	13	iic400kt5v	data input/output of I^2C-bus
V_{SSI1}	14	vssis	ground supply (core only)
V_{SSA2}	15	vssco	ground supply analog of PLL, WS_PLL, SPDIF input stage
V_{DDI1}	16	vddi	positive supply (core only) (bond out to 2 pads)
V_{DDA2}	17	vddco	positive supply analog of PLL, WS_PLL, SPDIF input stage
DSP_RESET	18	ipthut5v	general reset of chip (active LOW)
RTCB	19	ipthdt5v	asynchronous reset test control block, connect to ground (internal pull down)
SHTCB	20	ipthdt5v	shift clock test control block (internal pull down)
V_{SSE}	21	vsse	ground supply (peripheral cells only)
CLK_IN	22	iptht5v	system clock input
V_{DDE}	23	vdde	positive supply (peripheral cells only)
SPDIF2	24	apio	SPDIF2 data input (internally multiplexed with digital serial input 3)
SPDIF1	25	apio	SPDIF1 data input (internally multiplexed with digital serial input 2)
TSCAN	26	ipthdt5v	scan control active HIGH (internal pull down)
SYSCLK	27	bpt4mthdt5v	$n \times f_s$ output of SAA7715
IIS_OUT4	28	ops5c	data pin of digital serial output 4
IIS_OUT3	29	ops5c	data pin of digital serial output 3
IIS_OUT2	30	ops5c	data pin of digital serial output 2
IIS_OUT1	31	ops5c	data pin of digital serial output 1
IIS_WS	32	ops5c	word select output belonging to digital serial output 1 to 4
IIS_BCK	33	ops5c	bit clock output belonging to digital serial output 1 to 4
VOUTL	34	apio	analog left output pin.
V_{DDA1}	35	vddo	FSDAC positive supply voltage (bond out to 2 pads)
VOUTR	36	apio	analog right output pin
V_{SSA1}	37	vsso	FSDAC ground supply voltage (bond out to 2 pads)
VREFDA	38	apio	voltage reference pin of FSDAC
POM	39	apio	power-on mute pin of FSDAC
POWERDOWN	40	ipthdt5v	standby mode of chip

SYMBOL	PIN	PIN TYPE	DESCRIPTION
DIV_CLK_IN	41	ipthdt5v	divide the input frequency on pin CLK_IN by two
DSP_INOUT5	42	bpts5thdt5v	digital input/output flag of the DSP-core (F5 of the status register)
DSP_INOUT6	43	bpts5thdt5v	digital input/output flag of the DSP-core (F6 of the status register)
DSP_INOUT7	44	bpts5thdt5v	digital input/output flag of the DSP-core (F7 of the status register)

Table 1 Brief explanation of used pin types

PIN TYPE	EXPLANATION
apio	analog I/O pad cell; actually pin type vddco
bpts5thdt5v	43 MHz bidirectional pad; push-pull input; 3-state output; 5 ns slew rate control; TTL; hysteresis; pull-down; 5 V tolerant
bpts5tht5v	43 MHz bidirectional pad; push-pull input; 3-state output; 5 ns slew rate control; TTL; hysteresis; 5 V tolerant
bpt4mthdt5v	bidirectional pad; push-pull input; 3-state output; 4 mA output drive; TTL; hysteresis; pull-down; 5 V tolerant
iic400kt5v	I^2C-bus pad; 400 kHz I^2C-bus specification; 5 V tolerant
ipthdt5v	input pad buffer; TTL; hysteresis; pull-down; 5 V tolerant
iptht5v	input pad buffer; TTL; hysteresis; 5 V tolerant
ipthut5v	input pad buffer; TTL; hysteresis; pull-up; 5 V tolerant
ops5c	output pad; push-pull; 5 ns slew rate control; CMOS
op4mc	output pad; push-pull; 4 mA output drive
vddco	V_{DD} supply to core only
vdde	V_{DD} supply to peripheral only
vddi	V_{DD} supply to core only
vddo	V_{DD} supply to core only
vssco	V_{SS} supply to core only (vssco does not connect the substrate)
vsse	V_{SS} supply to peripheral only
vssi	V_{SS} supply to core and peripheral
vssis	V_{SS} supply to core only; with substrate connection
vsso	V_{SS} supply to core only

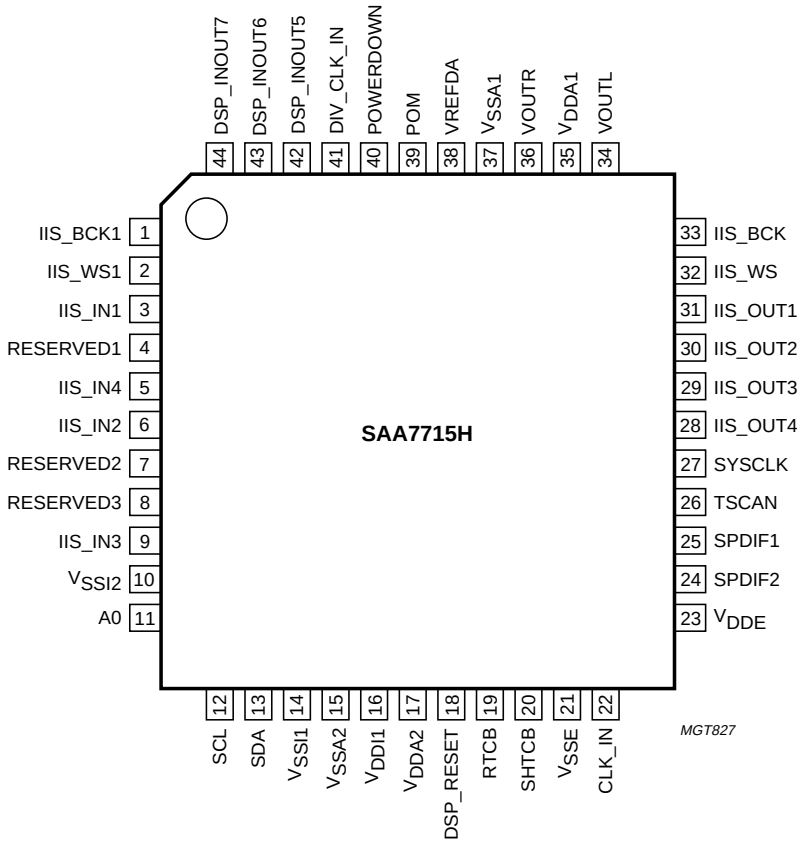


Fig.2 Pin configuration.

8 FUNCTIONAL DESCRIPTION

8.1 PLL division factors for different clock inputs

An on-chip PLL generates the clock for the DSP. The DSP runs at a selectable frequency of maximum 70 MHz. The clock is generated with the PLL that uses the CLK_IN of the chip to generate the DSP clock. Table 2 gives the division factors and the values of the DSP_TURBO and the DIV_CLK_IN bits that need to be set via I^{2}C-bus (see Table 10).

Table 2 PLL division factor per clock input.

CLOCK INPUT (MHz)	pll_div[4:0]	N	DSP_TURBO	DIV_CLK_IN	DSP_CLOCK (MHz)
8.192 (32 kHz × 256)	10H	272	1	0	69.632
9.728 (38 kHz × 256)	09H	227	1	0	69.008
11.2896 (44.1 kHz × 256)	03H	198	1	0	69.854
12.288 (48 kHz × 256)	00H	181	1	0	69.504
16.384 (32 kHz × 512)	10H	272	1	1	69.632
18.432 (32 kHz × 576)	0BH	244	1	1	68.544
19.456 (38 kHz × 512)	09H	227	1	1	69.008
24.576 (96 kHz × 256)	00H	181	1	1	69.504

The above table does NOT imply that the clock input is restricted to the values given in this table. The clock input is restricted to be within the range of 8.192 to 12.228 MHz. For higher clock frequencies pin DIV_CLK_IN should be set to logic 1 performing a divide by 2 of the CLK_IN signal and thereby doubling the CLK_IN frequency range that is allowed (16.384 to 24.576 MHz).

8.2 The word select PLL

A second on-chip PLL generates a selectable multiple of the sample rate frequency supplied on the word select pin IIS_WS (= IIS_WS1). The clock generated by this so called WS_PLL is available for the user at pin SYSCLK. Tables 3 and 4 show the I^{2}C-bus settings needed to generate the $n \times f_s$ clock. The memory map of the I^{2}C-bus bits is shown in Table 10.

Table 3 Word select input range selection

SAMPLE RATE OF f_s (kHz)	sel_loop_div[1:0]
32 to 50	01
50 to 96	00

Table 4 Selection of $n \times f_s$ clock at SYSCLK output

sel2	sel1	sel0	SYSCLK (n × IIS_WS1)	DUTY FACTOR
1	0	0	512	50% for 32 to 50 kHz input; 66% for 50 to 96 kHz input
0	1	1	384	50%
0	1	0	256	50%
0	0	1	192	50%
0	0	0	128	50%

8.3 The Filter Stream DAC (FSDAC)

The FSDAC is a semi-digital reconstruction filter that converts the 1-bit data stream of the noise shaper to an analog output voltage. The filter coefficients are implemented as current sources and are summed at virtual ground of the output operational amplifier. In this way very high signal-to-noise performance and low clock jitter sensitivity is achieved. A post-filter is not needed due to the inherent filter function of the DAC. On-board amplifiers convert the FSDAC output current to an output voltage signal capable of driving a line output.

The output voltage of the FSDAC scales proportionally with the power supply voltage.

8.3.1 Interpolation filter

The digital filter interpolates from 1 to 64f_s by means of a cascade of a recursive filter and an FIR filter.

Table 5 Digital interpolation filter characteristics

ITEM	CONDITIONS	VALUE (dB)
Pass band ripple	0 to 0.45f_s	±0.03
Stop band	>0.55f_s	–50
Dynamic range	0 to 0.45f_s	116.5
Gain	DC	–3.5

8.3.2 Noise shaper

The 5th-order noise shaper operates at 64f_s. It shifts in-band quantization noise to frequencies well above the audio band. This noise shaping technique enables high signal-to-noise ratios to be achieved. The noise shaper output is converted into an analog signal using a filter stream digital-to-analog converter.

8.3.3 Function of pin POM

With pin POM it is possible to switch off the reference current of the DAC. The capacitor on pin POM determines the time after which this current has a soft switch-on. So at power-on the current audio signal outputs are always muted. The loading of the external capacitor is done in two stages via two different current sources. The loading starts at a current level that is lower than the current loading after the voltage on pin POM has passed a particular level. This results in an almost dB-linear behaviour. This prevents ‘plop’ effects during power on/off.

8.3.4 Power off plop suppression

To avoid plops in a power amplifier, the supply voltage of the analog part of the DAC and the rest of the chip can be fed from a separate supply of 3.3 V. A capacitor connected to this supply enables to provide power to the analog part at the moment the digital voltage is switching off fast. In this event the output voltage will decrease gradually allowing the power amplifier some extra time to switch off without audible plops.

8.3.5 Pin VREFDA for internal reference

With two internal resistors half the supply voltage V_{DDA1} is obtained and used as an internal reference. This reference voltage is used as DC voltage for the output operational amplifiers and as reference for the DAC. In order to obtain the lowest noise and to have the best ripple rejection, a filter capacitor has to be added between this pin and ground, preferably close to the analog pin V_{SSA1}.

8.3.6 Supply of the analog outputs

The entire analog circuitry of the DACs and the OPAMPS are supplied by 2 supply pins, V_{DDA1} and V_{SSA1}. The V_{DDA1} must have sufficient decoupling to prevent THD degradation and to ensure a good Power Supply Rejection Ratio (PSRR). The digital part of the DAC is fully supplied from the chip core supply.

8.4 External control pins

The flags DSP_INOUT5 to DSP_INOUT7 are available as external pins. The flags can be used by the DSP depending on the downloaded software.

8.5 Digital serial inputs/outputs and SPDIF inputs

8.5.1 DIGITAL SERIAL INPUTS/OUTPUTS

For communication with external digital sources a digital serial bus is implemented. It is a serial 3-line bus, having one line for data, one line for clock and one line for the word select. For external digital sources the SAA7715 acts as a slave, so the external source is master and supplies the clock.

For the I²S-bus format itself see the official specification from Philips.

The digital serial input is capable of handling Philips I²S-bus and LSB-justified formats of 16, 18, 20 and 24 bits word sizes. The sampling frequency can be 32 up to 96 kHz. See the I²C-bus memory map for the bits that must be programmed, for selection of the desired serial format.

See Fig.3 for the general waveforms of the possible formats.

When the applied word length exceeds 24 bits, the LSBs are skipped.

The digital serial input/output circuitry is limited in handling the number of BCK pulses per WS period. The maximum allowed number of bit clocks per WS period is 256. Also the number of bit clocks during WS LOW and HIGH must be equal (50% WS duty factor) only for the LSB-justified formats.

There are two modes in which the digital inputs can be used (the mode is selectable via an I²C-bus bit):

- Use up to 4 digital serial inputs (8ch) with common WS and BCK signal (8ch IN and 8ch OUT + 2ch FSDAC output)
- Use one of the 2 SPDIF inputs as source instead of the use of the digital serial inputs (2ch IN and 8ch OUT + One 2ch FSDAC output).

8.5.2 SPDIF INPUTS

Two separate SPDIF receivers are available, one shared with digital serial input 2 (SPDIF1) and one with the digital serial input 3 (SPDIF2). The sample frequency at which the SPDIF inputs can be used must be in the range of 32 to 96 kHz.

There are few control signals available from the SPDIF input stage. These are connected to flags of the DSP:

- A lock signal indicating if the SPDIF input 1 or 2 is in lock
- The pcm_audio/non-pcm_audio bit indicating if an audio or data stream is detected on SPDIF input 1 or 2. The FSDAC output will NOT be muted in the event of non-audio PCM stream. This status bit can be read via the I²C-bus, the microprocessor controller can decide to put the DAC into MUTE (via pin POM).

Handling of channel status bits: The first 40 (of 192) channel status bits of the selected SPDIF source (0FFBH, bit 20), will come available in the I²C-bus registers 0FF2H to 0FF5H. Two registers 0FF2H to 0FF3H contain the information for the right channel, the other two (0FF4H to 0FF5H) contain the information for the left channel. The information can be read via I²C-bus or by the DSP program.

The design fulfils the digital audio interface specification *"IEC 60958-1 Ed2, part 1, general part IEC 60958-3 Ed2, part 3, consumer applications"*.

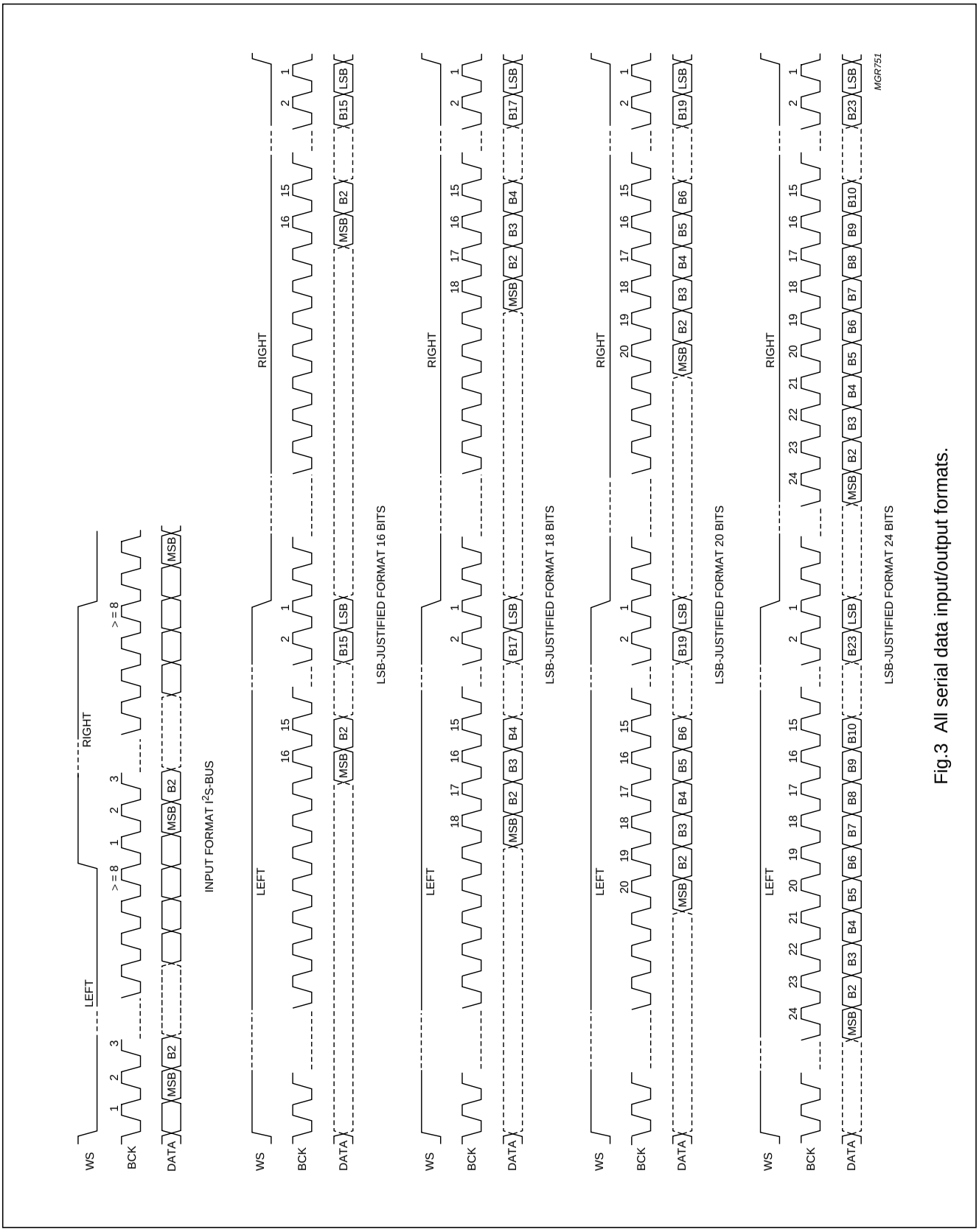


Fig.3 All serial data input/output formats.

8.6 I²C-bus interface (pins SCL and SDA)

The I²C-bus format is described in *"The I²C-bus and how to use it"*, order no. 9398 393 40011.

For the external control of the SAA7715 a fast I²C-bus is implemented. This is a 400 kHz bus which is downward compatible with the standard 100 kHz bus.

There are two different types of control instructions:

- Loading of the Program RAM (PRAM) with the required DSP program
  - Programming the coefficient RAM (YRAM)
  - Instructions to control the DSP program.
- Selection of the digital serial input/output format to be used, the DSP clock speed.

The detailed description of the I²C-bus and the description of the different bits in the memory map is given in Chapter 9.

8.7 Reset

The reset (pin DSP_RESET) is active LOW and needs an external 22 kΩ pull-up resistor. Between this pin and the V_{SSI} ground a capacitor of 1 μF should be connected to allow a proper switch-on of the supply voltage. The capacitor value is such that the chip is in reset as long as the power supply is not stabilized. A more or less fixed relationship between the DSP reset and the POM time constant is obligatory. The voltage on pin POM determines the current flowing in the DACs.

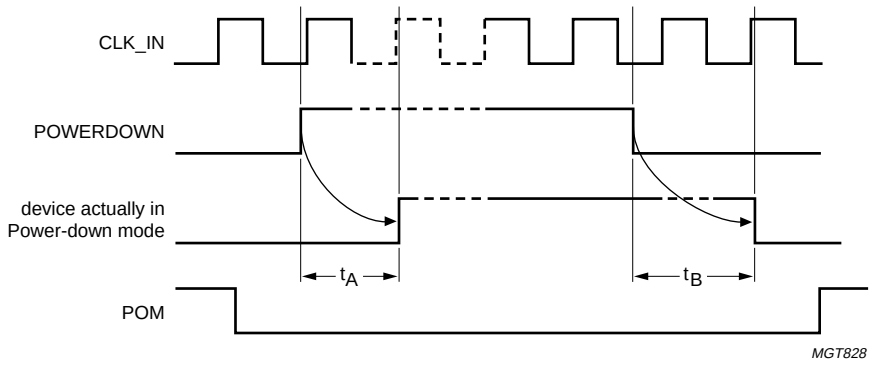
The reset sets all I²C-bus bits to their default value and it restarts the DSP program.

8.8 Power-down mode

The Power-down mode switches off all activity on the chip. The Power-down mode can be switched on and off using pin POWERDOWN. This pin needs to be connected to ground if not used. The following applies for the Power-down mode:

- Power-down mode may only be switched on when there is no I²C-bus activity to or from the SAA7715
- Power-down mode may not be switched on before the complete chip has been reset (DSP_RESET active LOW)
- The clock signal on pin CLK_IN should be running during Power-down mode
- It is advised to set pin POM to logic 0 before switching on the Power-down mode and set it back to logic 1 after the chip actually returns from Power-down mode as shown in Fig.4
- All on-chip registers and memories will keep their values during Power-down mode
- Digital serial outputs are not muted, the last value is kept on the output
- The SAA7715 will not 'lock-up' the I²C-bus during Power-down mode (SDA line).

Figure 4 shows the time the chip actually is in Power-down mode after switching on/off pin POWERDOWN.



$t_A = 4 \times (256/CLK_IN)$; 8.192 MHz < CLK_IN < 12.288 MHz.
$t_A = 4 \times (512/CLK_IN)$; 16.384 MHz < CLK_IN < 24.576 MHz.
$t_B = 128 \times (256/CLK_IN)$; 8.192 MHz < CLK_IN < 12.288 MHz.
$t_B = 128 \times (512/CLK_IN)$; 16.384 MHz < CLK_IN < 24.576 MHz.

Fig.4 Power-down mode.

8.9 Power supply connection and EMC

The digital part of the chip has in total 4 positive supply line connections and 5 ground connections. To minimize radiation the chip should be put on a double layer printed-circuit board with on one side a large ground plane. The ground supply lines should have a short connection to this ground plane. A coil/capacitor network in the positive supply line of the peripheral power supply line can be used as high frequency filter. The core supply lines (V_{DDI}) have an on-chip decoupling capacitance, for EMC reasons an external decoupling capacitance must not be used on this pin. A series resistor plus capacitance is required for proper operation on pin V_{DDA2}, see Fig.11.

8.10 Test mode connections (pins TSCAN, RTCB and SHTCB)

Pins TSCAN, RTCB and SHTCB are used to put the chip in test mode and to test the internal connections. Each pin has an internal pull-down resistor to ground. In the application these pins can be left open or connected to ground.

9 I²C-BUS PROTOCOL

9.1 Addressing

Before any data is transmitted on the I²C-bus, the device that should respond is addressed first. The addressing is always done with the first byte transmitted after the start procedure.

9.2 Slave address (pin A0)

The SAA7715 acts as slave receiver or a slave transmitter. Therefore the clock signal SCL is only an input signal. The data signal SDA is a bidirectional line. The slave address is shown in Table 6.

Table 6 Slave address

MSB							LSB
0	0	1	1	1	1	A0	R/$\overline{W}$

The sub-address bit A0 corresponds to the hardware address pin A0 which allows the device to have 2 different addresses. The A0 input is also used in test mode as serial input of the test control block.

9.3 Write cycles

The I²C-bus configuration for a write cycle is shown in Fig.5. The write cycle is used to write the bytes to the DSP for manipulating the data and coefficients. More details can be found in the I²C-bus memory map, see Table 8.

The data length is 2, 3 or 4 bytes depending on the accessed memory. If the Y-memory is addressed the data length is 2 bytes, in the event of the X-memory the length is 3 bytes. The slave receiver detects the address and adjusts the number of bytes accordingly.

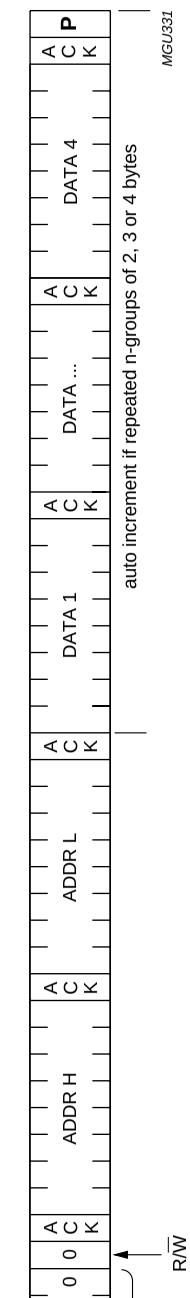
For this RAM-based product the internal P-memory (PRAM) can be accessed via the I²C-bus interface. The transmitted data-stream should be 4 bytes.

9.4 Read cycles

The I²C-bus configuration for a read cycle is shown in Fig.6. The read cycle is used to read the data values from XRAM, YRAM or PRAM. The master starts with a START condition S, the SAA7715 address ‘0011110’ and a logic 0 (write) for the read/write bit. This is followed by an acknowledge of the SAA7715. Then the master writes the high memory address (ADDR H) and low memory address (ADDR L) where the reading of the memory content of the SAA7715 must start. The SAA7715 acknowledges these addresses both.

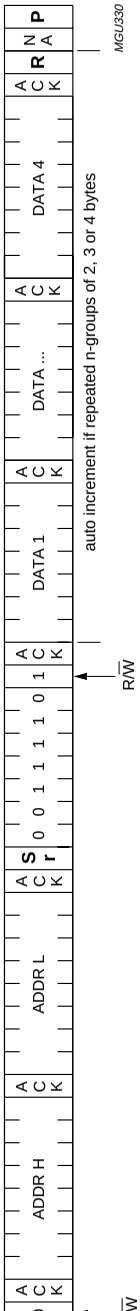
The master generates a repeated START (Sr) and again the SAA7715 address ‘0011110’ but this time followed by a logic 1 (read) of the read/write bit. From this moment on the SAA7715 will send the memory content in groups of 3 (X/Y-memory or registers) or 4 (P-memory) bytes to the I²C-bus each time acknowledged by the master. The master stops this cycle by generating a negative acknowledge, then the SAA7715 frees the I²C-bus and the master can generate a STOP condition.

The data is transferred from the DSP register to the I²C-bus register at execution of the MPI instruction in the DSP program. Therefore at least once every DSP routine an MPI instruction should be added.



S = START condition.
P = STOP condition.
ACK = acknowledge from SAA7715.
ADDR H and ADDR L = address DSP register.
DATA 1 to DATA 4 = 2, 3 or 4 bytes data word.

Fig.5 Master transmitter writes to the SAA7715 registers.



S = START condition.
Sr = repeated START condition.
P = STOP condition.
ACK = acknowledge from SAA7715 (SDA LOW).
R = repeat n-times the 2, 3 or 4 bytes data group.
NA = Negative Acknowledge master (SDA HIGH).
ADDR H and ADDR L = address DSP register.
DATA 1 to DATA 4 = 2, 3 or 4 bytes data word.

Fig.6 Master transmitter reads from the SAA7715 registers.

9.5 Program RAM

The SAA7715 has a 1.5 kbyte PRAM to store the DSP instruction code into. Also a 2 kbyte PROM is on-chip available and can be accessed (memory mapped) without the need of selecting the PROM or PRAM. The DSP instruction code can be downloaded into the PRAM via the I^{2}C-bus. The write and read cycle are shown in Figs 5 and 6 respectively.

The DSP has an instruction word width of 32 bits which means that this space should be accessed with 4 bytes in consecutive order and does have the auto-increment function.

9.6 Data word alignment

It is possible to transfer data via the I^{2}C-bus to a destination where it can have different data word length. Those destinations with data word are shown in Table 7.

Table 7 Data word alignment

SOURCE	DESTINATION	DATA WORD	BYTES (NUMBER)
I^{2}C-bus	DSP-PRAM	MBBB BBBB BBBB BBBB BBBB BBBB BBBB BBBL	4
I^{2}C-bus	DSP and general control	MBBB BBBB BBBB BBBB BBBB BBBL	3
I^{2}C-bus	I^{2}C-bus registers	MBBB BBBB BBBB BBBB BBBB BBBL	3
I^{2}C-bus	DSP-XRAM	MBBB BBBB BBBB BBBB BBBB BBBL	3
I^{2}C-bus	DSP-YRAM	XXXX MBBB BBBB BBBL	2

9.7 I²C-bus memory map specification

The I²C-bus memory map contains all defined I²C-bus bits. The map is split up in two different sections: the hardware memory registers and the RAM definitions. In Table 8 the preliminary memory map is depicted. The hardware registers are memory map on the XRAM of DSP. Table 9 shows the detailed memory map of those locations. All locations are acknowledged by the SAA7715 even if the user tries to write to a reserved space. The data in these sections will be lost. Reading from these locations will result in undefined data words.

Table 8 I²C-bus memory map

ADDRESS	FUNCTION	SIZE
8000H to 87FFH	DSP to PROM (not readable via I²C-bus)	2k × 32 bits
602FH	DSP and general control	1 × 24 bits
2000H to 25FFH	DSP to PRAM	1.5k × 32 bits
1000H to 01FFH	DSP to YRAM	512 × 12 bits
0FF2H to 0FF5H, 0FFBH	I²C-bus register	1 × 24 bits
0000H to 09FFH	DSP to XRAM	2.5k × 24 bits

Table 9 I²C-bus memory map overview

ADDRESS	DESCRIPTION
Hardware registers	
0FFBH	Selector register 1
0FF5H	SPDIF IN channel status register 1 left
0FF4H	SPDIF IN channel status register 2 left
0FF3H	SPDIF IN channel status register 1 right
0FF2H	SPDIF IN channel status register 2 right
DSP control	
602FH	DSP and general control register

9.8 I²C-bus memory map definition

Table 10 DSP and general control register (602FH)

NAME	SIZE (BITS)	DESCRIPTION	DEFAULT	BIT POSITION
	1	reserved	0	0
pll_div[4:0]	5	PLL clock division factor according to Table 2	00011	5 to 1
dsp_turbo	1	PLL output frequency 1: double 0: no doubling	1	6
	1	reserved	1	7
pc_reset_dsp	1	program counter reset DSP 1: reset on 0: reset off	0	8
	2	reserved	00	10 to 9
sel[2:0]	3	selection of $n \times f_s$ clock at SYSCLK output according to Table 4	010	13 to 11
sel_loop_div[1:0]	2	word select input range selection for WS_PLL according to Table 3	01	15 to 14
	2	reserved	00	17 to 16
sel_FSDAC_clk	2	clock source for FSDAC 00: WS_PLL if no signal to pin CLK_IN 01: $512f_s$ to pin CLK_IN 11: $256f_s$ to pin CLK_IN	00	19 to 18
dis_SYSCLK	1	output on pin SYSCLK 1: disable 0: enable	0	20
$256f_s$_n*Fs	1	signal on pin SYSCLK 1: fixed $256f_s$ clock 0: $n \times f_s$ clock; determined by bits 13 to 11	0	21
	1	reserved	0	23 to 22

Table 11 SPDIF IN channel status register 2 right (0FF2H)

NAME	SIZE (BITS)	DESCRIPTION	DEFAULT	BIT POSITION
ch_stat_in right lsb	20	channel status SPDIF in right LSB bits 19 to 0	00000H	19 to 0

Table 12 SPDIF IN channel status register 1 right (0FF3H)

NAME	SIZE (BITS)	DESCRIPTION	DEFAULT	BIT POSITION
ch_stat_in right msb	20	channel status SPDIF in right MSB bits 39 to 20	00000H	19 to 0

Table 13 SPDIF IN channel status register 2 left (0FF4H)

NAME	SIZE (BITS)	DESCRIPTION	DEFAULT	BIT POSITION
ch_stat_in left lsb	20	channel status SPDIF in2 left LSB bits 19 to 0	00000H	19 to 0

Table 14 SPDIF IN channel status register 1 left (0FF5H)

NAME	SIZE (BITS)	DESCRIPTION	DEFAULT	BIT POSITION
ch_stat_in left msb	20	channel status SPDIF in2 left MSB bits 39 to 20	00000H	19 to 0

Table 15 Selector register 1 (0FFBH)

NAME	SIZE (BITS)	DESCRIPTION	DEFAULT	BIT POSITION
format_in1	3	digital serial inputs 1 and 4 data format according to Table 17	011	2 to 0
format_in2	3	digital serial input 2 data format according to Table 17	011	5 to 3
format_in3	3	digital serial input 3 data format according to Table 17	011	8 to 6
format_out	3	digital serial outputs 1 to 4 data format according to Table 18	000	11 to 9
en_output	1	enable or disable digital serial outputs 1: enable 0: disable	1	12
	1	reserved	0	13
master_source	4	source selection 0000: digital serial input 1 0101: digital serial input 2 or SPDIF 1 (see bit 18) 1010: digital serial input 3 or SPDIF 2 (see bit 19) all other values are reserved	0000	14 to 17
spdif_sel1	1	SPDIF1 or digital serial input 2	0	18
		1: SPDIF1		
		0: digital serial input 2		
spdif_sel2	1	SPDIF2 or digital serial input 3	0	19
		1: SPDIF2		
		0: digital serial input 3		
sel_spdifin_chstat	1	select channel status information taken from SPDIF1 or SPDIF2	0	20
		1: from input SPDIF2		
		0: from input SPDIF1		
	3	reserved	000	21 to 23

Table 16 Default settings of I²C-bus registers after power-up and reset

I²C-BUS ADDRESS	DEFAULT VALUE
602FH	0050C6H
0FFBH	0010DBH
0FF5H	000000H
0FF4H	000000H
0FF3H	000000H
0FF2H	000000H

9.9 Table definitions

Table 17 Digital serial format for inputs 1 to 4

FORMAT_IN 1, 2 AND 3			OUTPUT
BIT 2	BIT 1	BIT 0	
0	1	1	standard I²S-bus
1	0	0	LSB-justified, 16 bits
1	0	1	LSB-justified, 18 bits
1	1	0	LSB-justified, 20 bits
1	1	1	LSB-justified, 24 bits

Table 18 Digital serial formats for outputs 1 to 4

FORMAT_OUT			OUTPUT
BIT 2	BIT 1	BIT 0	
0	0	0	standard I²S-bus
1	0	0	LSB-justified, 16 bits
1	0	1	LSB-justified, 18 bits
1	1	0	LSB-justified, 20 bits
1	1	1	LSB-justified, 24 bits

10 SOFTWARE IN ROM DESCRIPTION

10.1 Audio dynamics compressor

10.1.1 THEORY OF OPERATION

The objective of a dynamics compressor is to reduce the dynamic range of the input signal for purposes of accommodating downstream devices, or simply to give the audio signal a different character. Early compressors were used primarily for limiting signals destined for recording on media with limited dynamic range. In the present day, compressors are routinely used in recording studios and in live performances to enhance the presence of various signals.

The behaviour of a dynamics compressor is very similar to that of an Automatic Gain Control (AGC), the central idea being to scale the input signal by a slowly varying gain factor that is in turn regulated by the level of the input signal. The essential concepts are summed up nicely by Fig.7. Here we observe that when the input level exceeds a selected threshold, gain reduction is brought to bear according to the selected compression ratio, while signals appearing below the threshold are passed with unity gain. The net effect, therefore, is to compress the louder passages of source material.

Ironically, most people think in terms of boosting the low signals when talking about dynamics compression. In fact, this is what actually happens after the output is rescaled to account for the gain reduction imposed by the current settings. By doing this the output signal can be forced to carry more power than the input. This is what gives the compressor its 'punch' quality, for a more 'in your face' sort of sound. Figure 8 shows an example of the transfer curves before and after application of output gain. Users should be aware, however, that abuse of output gain can amplify system noise to intolerable levels.

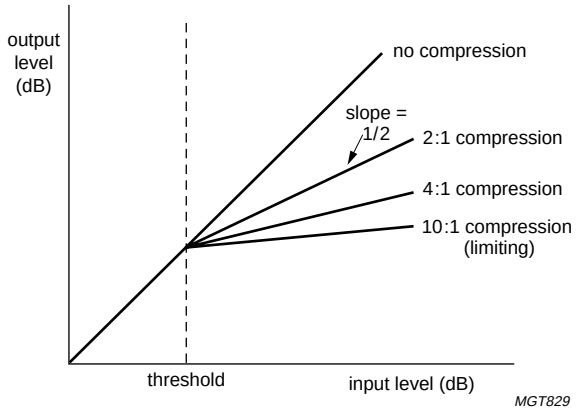


Fig.7 Gain reduction is applied only when the signal exceeds the set threshold level.

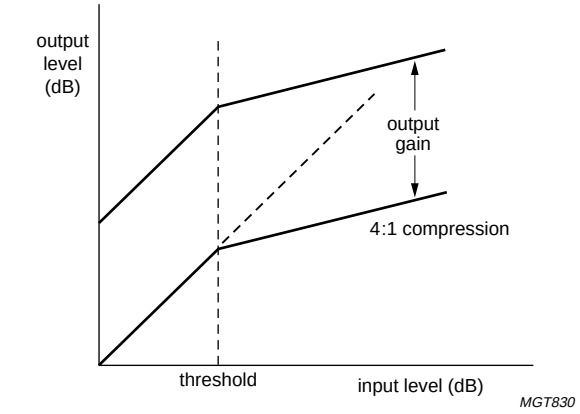


Fig.8 Output gain can be used to restore the peak level to its maximum.

10.1.1.1 Control parameters

Common to most compressors are five control parameters used for adjusting the behaviour of the compressor. These are typically labelled as threshold, ratio, attack time, release time, and output. By careful adjustment of these controls a skilled user can produce very pleasing results for a wide variety input source material. In the following subsections, functionality of each control is described.

10.1.1.2 Fixed versus variable mode

The compressor module can be operated in so-called 'fixed' mode or 'variable' mode. When in variable mode, the user has full control over both the threshold and ratio controls. In fixed mode, controls are frozen and the effect operates at a fixed ratio of 2:1, with a threshold setting of –36 dB(FS). These settings were chosen as a good compromise for a wide variety of source material.

10.1.1.3 Threshold

Threshold determines the level at which gain reduction begins. For example, if the threshold is set at –10 dB(FS), this means that all signals below –10 dB(FS) will be passed unaltered. Only when the input level exceeds this threshold is gain reduction (compression) brought to bear.

Many times a dramatic change in the threshold setting will call for a ratio adjustment. Experiment with these two controls to find what works best for your system, your music, and most importantly, your ears.

10.1.1.4 Ratio

The ratio control sets the desired compression ratio. Settings are traditionally expressed in ratios such as 1.5:1, 2:1, 4:1, 10:1, etc. An explanation of how to interpret these settings is best served by example. Say we are dealing with a ratio of 1.5:1. This means that for every 1.5 dB increase in input level beyond the threshold, only 1 dB is passed to the output. Another way of explaining this is in terms of gain reduction. In this particular case a 0.5 dB gain reduction is imposed for every 1.5 dB increase beyond the threshold level.

Compression ratio is changed by selecting one of the values in the drop-down list labelled 'Ratio'. To increase the amount of compression, select one of the higher ratios. For a more subtle effect, select a lower setting, such as 1.5:1.

10.1.1.5 Attack time

Attack time controls the rate at which gain-reduction is engaged following the detection of the input signal exceeding the threshold level. Typical values are in the range of 0 to 100 ms. Fast attack times tend to smooth out abrupt transients thereby helping to ensure the output level remains fairly consistent; however, at the same time fast attack times can easily destroy much of the dynamic character of sources having very distinguished attack transients (such as a piano or an acoustic guitar). Slow attack times, on the other hand, allow the sources attack transients to pass through virtually unaltered, thereby retaining most of the dynamic signature of the source. The danger here, however, is the possibility of clipping the output, or overloading one or more downstream components.

The present implementation of the compressor does not provide user access to attack time.

10.1.1.6 Release time

Complementing the attack time control, release time controls the speed at which the compressor disengages after the input level falls back below the threshold. Typical values here range from around 100 ms to several seconds.

The present implementation of the compressor does not provide user access to release time.

10.1.1.7 Output or 'make-up gain'

In order to make maximum use of the available bit resolution, it becomes necessary to boost the compressors output in order to ensure the signal swings close to the maximum excursions allowed by the digital output. Notice in Fig.7 how the output level can be dramatically reduced, particularly at low threshold levels and high compression ratios. In the present implementation, this rescaling is managed automatically according to the current threshold and ratio settings.

10.2 Audio enhancer

10.2.1 THEORY OF OPERATION

The enhancer uses non-linear processing to generate extra harmonics, which are added to the audio to improve high frequency detail. It is particularly useful with streaming audio from the Internet, which is typically compressed to the extent that the original high frequency content is lost.

The enhancer is also a very effective means of improving the sound of CD-quality audio, by restoring the presence and brilliance of the original acoustic performance.

10.2.1.1 Control parameters

The enhancer has a single mix control, which determines the amount of generated harmonics to be added to the signal. High settings will result in a brighter effect with greater depth. For particularly dull audio, such as is often received over the Internet, a high mix level will have a pleasing effect. Intermediate settings are appropriate for CD-quality audio, although classical music listeners may prefer to use the enhancer sparingly.

10.3 Equalizer

10.3.1 GENERAL DESCRIPTION

- 2-channels
- 5-bands
- Control range: 20 Hz to 20 kHz.

10.3.2 OVERVIEW

The fundamental ideal for any high-fidelity audio rendering system is to reproduce the aural experience present at the time and place the original audio material was recorded. Unfortunately, practically all systems fall short of this ideal to some degree for a number of reasons. While environmental acoustics can play a significant role, in many cases performance deficiencies associated with the loudspeakers cause most of the 'distortion'. This happens when the loudspeakers cannot deliver a uniform frequency response over the entire audio range (20 Hz to 20 kHz).

Equalizers were invented to deal with frequency response problems by boosting or cutting selected frequency bands in the signal. Used in the right manner, a properly adjusted equalizer can effectively compensate for loudspeaker performance deficiencies, or any other frequency dependent amplitude variations in the system. Additionally, equalization can be used to create a customized frequency response which is better suited for a particular listener or a particular style of music, for instance.

The type of equalizer provided with this system is of the parametric variety. Parametric equalizers differ from graphic equalizers by giving the user more control over the filters that actually effect the boost or cut of a particular band. More specifically, for each band, users can control the band's centre frequency, and also the width of the band of frequencies that are affected.

10.3.3 CONTROLS

The equalizer module exposes three controls for each of the five bands. These are referred to as gain, centre frequency, and Q. The gain control sets the amount of boost or cut applied to the particular band of frequencies. Centre frequency controls the frequency at which the boost or cut filter is centred, while the Q control determines the bandwidth (the range of frequencies) over which the filter operates.

10.3.4 CENTRE FREQUENCY

Centre frequency defines the frequency where boost or cut will be centred. To set the centre frequency, select the entry box and type in a number that is within the allowed range.

10.3.5 GAIN

Use the gain control to adjust the amount or boost or cut. Move the slider upward (above the 0 dB line) to add boost, downward (below the 0 dB line) to cut.

10.3.6 Q

The Q parameter determines the sharpness of the filter. As the value of Q increases, the filter becomes narrower, thereby reducing the filters effective bandwidth. High Q filters are useful for reducing speaker resonances, or for eliminating resonance that may be caused by the acoustic environment. Low Q filters, on the other hand, are useful for operating on a broad range of frequencies.

10.3.7 HINTS AND TIPS

Avoid using the equalizer for volume control. This is not the purpose of an equalizer. Remember, you are only trying to correct frequency response deviations from some ‘ideal’ response that are due to loudspeaker deficiencies and perhaps the surrounding environment. Therefore you should strive to introduce the minimum amount of equalization that causes the system output to reach your desired response.

Avoid excessive boost or cut. This can introduce noticeable coloration of the program material.

10.4 Stereo spatializer

10.4.1 OVERVIEW

In PC listening settings, the quality of the stereo image is sometimes compromised by the short distance between the loudspeakers, and also by the physical limitations of the loudspeakers themselves. The spatializer effect remedies these shortcomings by applying perceptually tuned signal-processing to create the illusion of a wider and more enveloping sound stage.

Users should be relieved to know that relative positioning of instruments in the original material is preserved. In other words, tracks that are centre mixed in the original material remain centred; tracks panned left or right in the original mix remain left and right panned. The main difference is in the apparent width and depth of the sound stage, it is as though the listener is hearing a larger and more distant pair of speakers, spaced much farther apart than those actually present.

10.4.2 CONTROLS

The spatializer effect uses only one control to change its behaviour.

10.4.3 MIX

The mix control sets the intensity of the effect. Control is straight-forward. Add more effect by pulling the slider upward; move the slider downward to reduce the amount of effect.

10.4.4 HINTS AND TIPS

Try a mix setting of about 0.7 as a starting point.

For best results, position yourself between the speakers and a couple of feet back. Ideally, your ears should be at about the same level as the speakers, but this is not so critical.

11 LIMITING VALUES

In accordance with the Absolute Maximum Ratings System (IEC 60134).

SYMBOL	PARAMETER	CONDITIONS	MIN.	MAX.	UNIT
V_{DD}	supply voltage		–0.5	+3.6	V
V_I	input voltage		–0.5	+5.5	V
I_{IK}	input clamping diode current	$V_I < -0.5$ V or $V_I > V_{DD} + 0.5$ V	–	±10	mA
I_{OK}	output clamping diode current	$V_O < -0.5$ V or $V_O > V_{DD} + 0.5$ V	–	±20	mA
$I_{O(sink/source)}$	output source or sink current	-0.5 V $< V_O < V_{DD} + 0.5$ V	–	±20	mA
I_{DD},I_{SS}	V_{DD} or V_{SS} current per supply pin		–	±50	mA
T_{amb}	ambient temperature		–40	+85	°C
T_{stg}	storage temperature		–65	+125	°C
V_{ESD}	electrostatic handling voltage	note 1	200	–	V
		note 2	2000	–	V
$I_{lu(prot)}$	latch-up protection current	CIC specification/test method	100	–	mA
P_{tot}	total power dissipation		–	600	mW

Notes

1. Machine model (R = 0 Ω; C = 100 pF; L = 2.5 μH).
2. Human body model (R = 1500 Ω; C = 100 pF).

12 THERMAL CHARACTERISTICS

SYMBOL	PARAMETER	CONDITIONS	VALUE	UNIT
$R_{th(j-a)}$	thermal resistance from junction to ambient	mounted on printed-circuit board	60	K/W

13 CHARACTERISTICS

V_{DD} = 3.15 to 3.45 V; unless otherwise specified.

SYMBOL	PARAMETER	CONDITIONS	MIN.	TYP.	MAX.	UNIT
Supplies; T_{amb} = −40 to +85 °C						
V_{DD}	operating supply voltage	all pins V_{DD} with respect to pins V_{SS}	3.15	3.3	3.45	V
I_{DDD}	supply current of the digital part		–	95	–	mA
$I_{DDD(core)}$	supply current of the digital core part	high activity of the DSP at DSPFREQ frequency	–	90	–	mA
$I_{DDD(peri)}$	supply current of the digital periphery part	no external load to ground	–	5	–	mA
I_{DDA}	supply current of the analog part	zero input and output signal	–	20	–	mA
$I_{DDA(DAC)}$	supply current of the DAC	zero input and output signal	–	6.5	13	mA
		Power-down mode	–	250	–	µA
$I_{DDA(SPDIF)}$	supply current of the SPDIF inputs, on-chip PLL and WSPLL	zero input and output signals	–	13.5	27	mA
P_{tot}	total power dissipation		–	380	–	mW
$I_{POWERDOWN}$	DC supply current of the total chip in Power-down mode	pin POWERDOWN enabled	–	400	–	µA
Digital I/O; T_{amb} = −40 to +85 °C; V_{DD} = 3.15 to 3.45 V; unless otherwise specified						
V_{IH}	HIGH-level input voltage all digital inputs and I/Os		2.0	–	–	V
V_{IL}	LOW-level input voltage all digital inputs and I/Os		–	–	0.8	V
V_{hys}	Schmitt-trigger hysteresis		0.4	–	–	V
V_{OH}	HIGH-level output voltage	standard output; I_O = –4 mA	V_{DD} – 0.4	–	–	V
		5 ns slew rate output; I_O = –4 mA	V_{DD} – 0.4	–	–	V
		10 ns slew rate output; I_O = –2 mA	V_{DD} – 0.4	–	–	V
		20 ns slew rate output; I_O = –1 mA	V_{DD} – 0.4	–	–	V

SYMBOL	PARAMETER	CONDITIONS	MIN.	TYP.	MAX.	UNIT
V_{OL}	LOW-level output voltage	standard output; I_O = 4 mA	–	–	0.4	V
		5 ns slew rate output; I_O = 4 mA	–	–	0.4	V
		10 ns slew rate output; I_O = 2 mA	–	–	0.4	V
		20 ns slew rate output; I_O = 1 mA	–	–	0.4	V
		I^2C-bus output; I_O = 4 mA	–	–	0.4	V
I_{LO}	output leakage current 3-state outputs	V_O = 0 V or V_{DD}	–	–	±5	µA
R_{pd}	internal pull-down resistor to V_{SS}		24	50	140	kΩ
R_{pu}	internal pull-up resistor to V_{DD}		30	50	100	kΩ
C_i	input capacitance		–	–	3.5	pF
$t_{i(r)}$,$t_{i(f)}$	input rise and fall times	V_{DD} = 3.45 V	–	6	200	ns
$t_{o(t)}$	output transition time	standard output; C_L = 30 pF	–	3.5	–	ns
		5 ns slew rate output; C_L = 30 pF	–	5	–	ns
		10 ns slew rate output; C_L = 30 pF	–	10	–	ns
		20 ns slew rate output; C_L = 30 pF	–	20	–	ns
		I^2C-bus output; C_L = 400 pF	60	–	300	ns
AC characteristics SPDIF1 and SPDIF2 inputs; T_{amb} = 25 °C; V_{DDA2} = 3.3 V; unless otherwise specified						
$V_{i(p-p)}$	AC input level (peak-to-peak level)		0.2	0.5	3.3	V
R_i	input impedance	at 1 kHz	–	6	–	kΩ
V_{hys}	hysteresis of input voltage		–	40	–	mV

SYMBOL	PARAMETER	CONDITIONS	MIN.	TYP.	MAX.	UNIT
Analog DAC outputs; V_{DDA1} = 3.3 V; f_s = 44.1 kHz; T_{amb} = 25 °C; R_L = 5 kΩ; all voltages referenced to ground; unless otherwise specified						
DC CHARACTERISTICS						
$R_{o(DAC)}$	DAC output resistance	pins 34 and 36	–	0.13	3.0	Ω
$I_{o(max)}$	maximum output current	(THD + N)/S < 0.1% R_L = 5 kΩ	–	0.22	–	mA
R_L	load resistance		3	–	–	kΩ
C_L	load capacitance		–	–	200	pF
$R_{o(VREFDA)}$	VREFDA output resistance	pin 38	–	28	–	kΩ
AC CHARACTERISTICS						
$V_{o(rms)}$	output voltage (RMS value)		–	1000	–	mV
ΔV_o	unbalance between channels		–	0.1	–	dB
(THD + N)/S	total harmonic distortion plus noise-to-signal ratio	at 0 dB	–	–85	–	dB(A)
		at –60 dB	–	–37	–	dB(A)
S/N	signal-to-noise ratio	code = 0	–	100	–	dB(A)
α_{cs}	channel separation		–	80	–	dB
PSRR	power supply rejection ratio	f_{ripple} = 1 kHz; $V_{ripple(p-p)}$ = 1%	–	50	–	dB

14 I²S-BUS TIMING

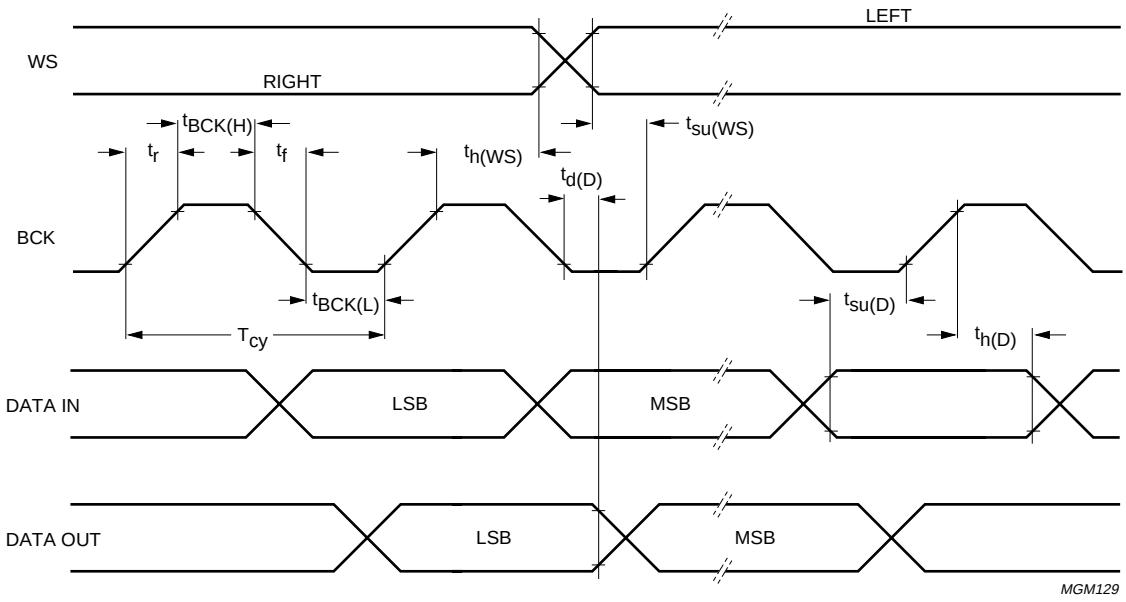


Fig.9 Timing of the digital audio data inputs and outputs.

Table 19 Timing digital serial audio inputs and outputs (see Fig.9)

SYMBOL	PARAMETER	CONDITIONS	MIN.	TYP.	MAX.	UNIT
T_{cy}	bit clock cycle time		162	–	–	ns
t_r	rise time	T_{cy} = 50 ns	–	–	$0.15T_{cy}$	ns
t_f	fall time	T_{cy} = 50 ns	–	–	$0.15T_{cy}$	ns
$t_{BCK(H)}$	bit clock HIGH time	T_{cy} = 50 ns	$0.35T_{cy}$	–	–	ns
$t_{BCK(L)}$	bit clock LOW time	T_{cy} = 50 ns	$0.35T_{cy}$	–	–	ns
$t_{su(D)}$	data set-up time	T_{cy} = 50 ns	$0.2T_{cy}$	–	–	ns
$t_{h(D)}$	data hold time	T_{cy} = 50 ns	$0.2T_{cy}$	–	–	ns
$t_{d(D)}$	data delay time	T_{cy} = 50 ns	–	–	$0.15T_{cy}$	ns
$t_{su(WS)}$	word select set-up time	T_{cy} = 50 ns	$0.2T_{cy}$	–	–	ns
$t_{h(WS)}$	word select hold time	T_{cy} = 50 ns	$0.2T_{cy}$	–	–	ns

15 I²C-BUS TIMING

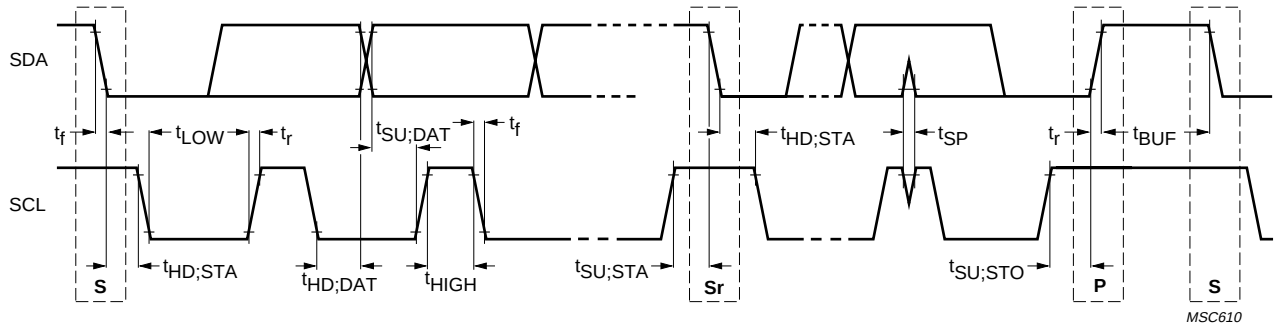


Fig.10 Definition of timing on the I²C-bus.

Table 20 Timing of I²C-bus (see Fig.10)

SYMBOL	PARAMETER	CONDITIONS	STANDARD MODE I²C-BUS		FAST MODE I²C-BUS		UNIT
			MIN.	MAX.	MIN.	MAX.	
f_{SCL}	SCL clock frequency		0	100	0	400	kHz
t_{BUF}	bus free time between a STOP and START condition		4.7	–	1.3	–	µs
$t_{HD;STA}$	hold time (repeated) START condition; after this period, the first clock pulse is generated		4.0	–	0.6	–	µs
t_{LOW}	SCL LOW period		4.7	–	1.3	–	µs
t_{HIGH}	SCL HIGH period		4.0	–	0.6	–	µs
$t_{SU;STA}$	set-up time for a repeated START condition		4.7	–	0.6	–	µs
$t_{HD;DAT}$	DATA hold time		0	–	0	0.9	µs
$t_{SU;DAT}$	DATA set-up time		250	–	100	–	ns
t_r	rise time of both SDA and SCL signals	C_b in pF	–	1000	$20 + 0.1C_b$	300	ns
t_f	fall time of both SDA and SCL signals	C_b in pF	–	300	$20 + 0.1C_b$	300	ns
$t_{SU;STO}$	set-up time for STOP condition		4.0	–	0.6	–	µs
C_b	capacitive load for each bus line		–	400	–	400	pF
t_{SP}	pulse width of spikes to be suppressed by input filter		not applicable		0	50	ns

16 APPLICATION DIAGRAM

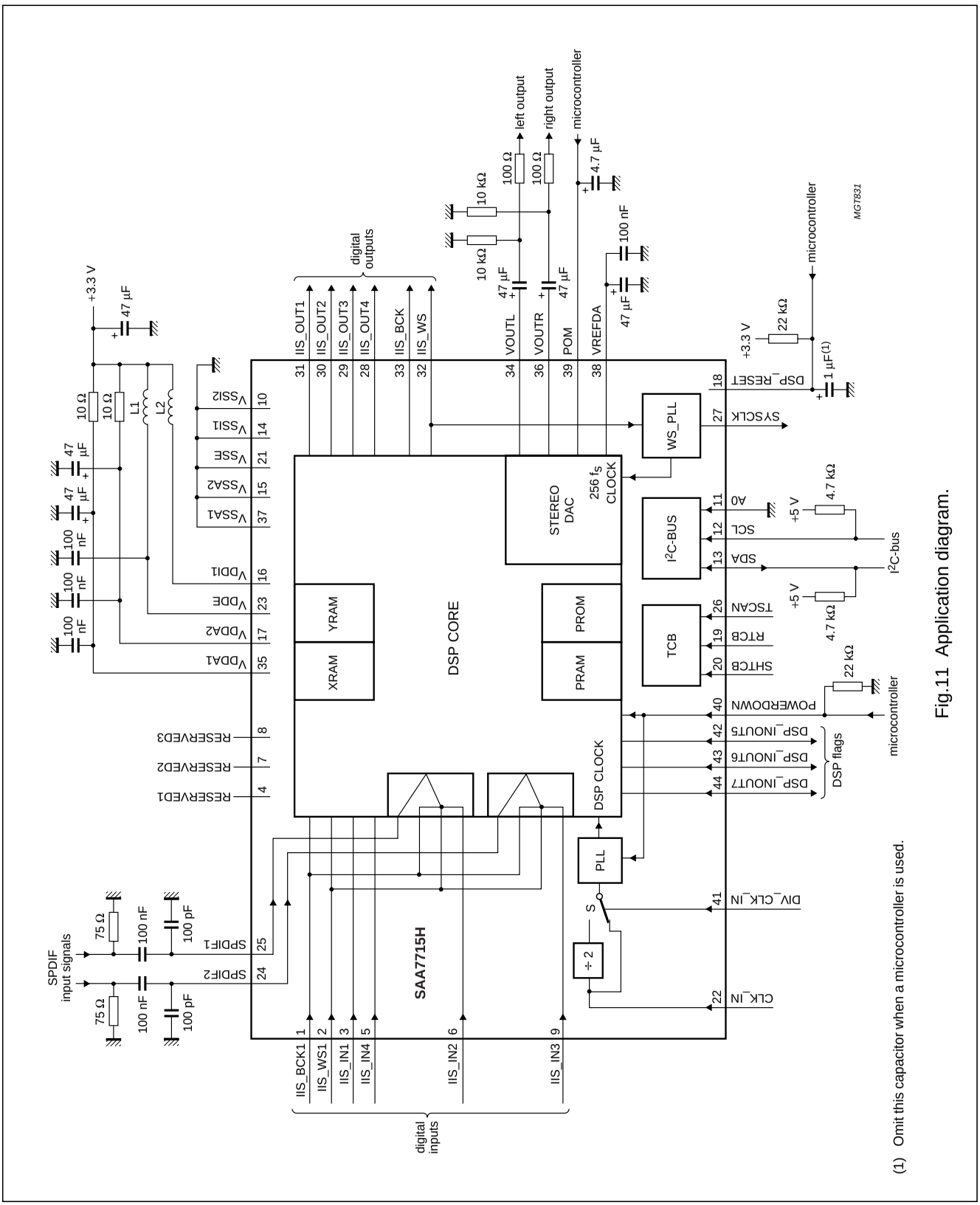
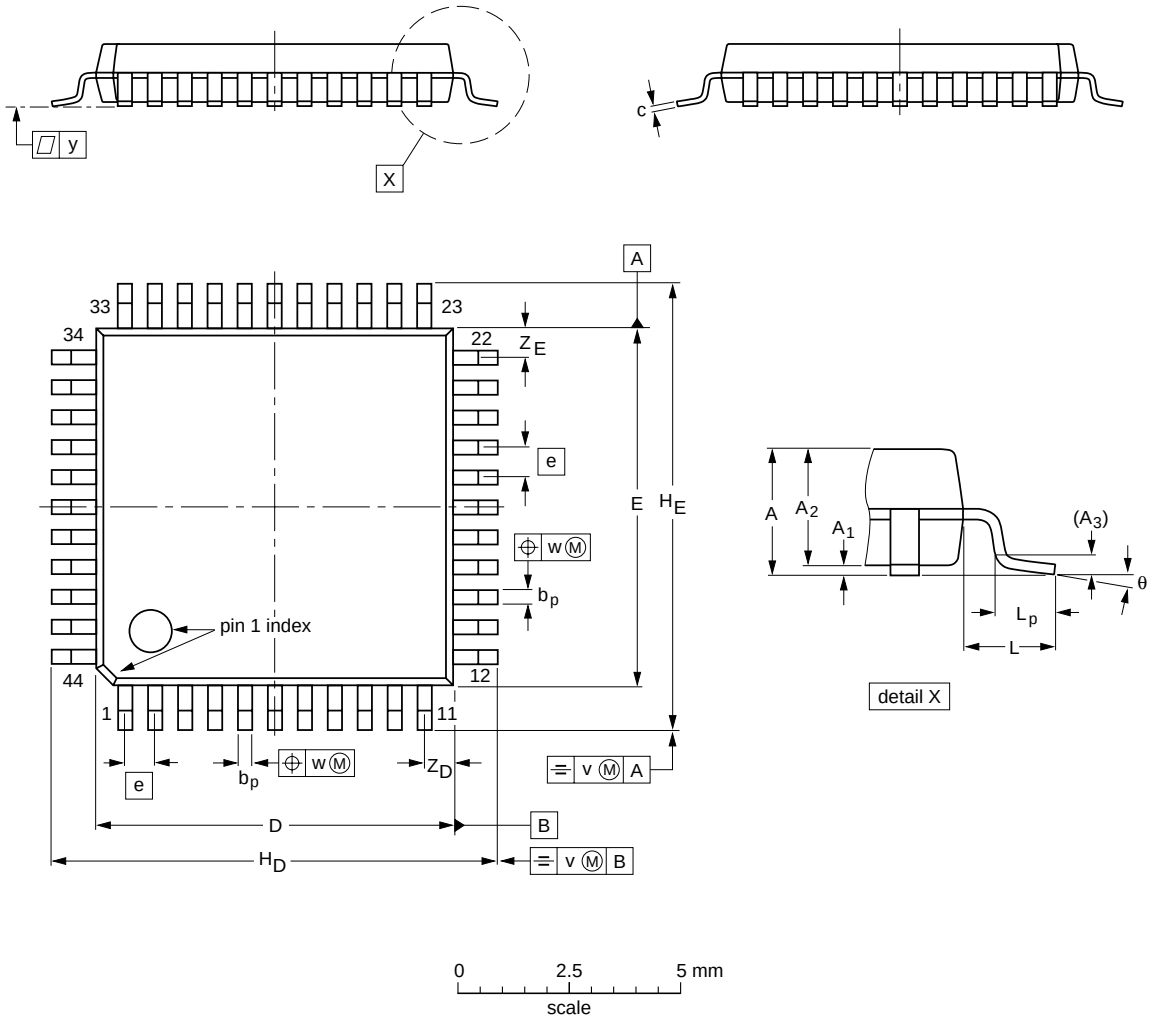


Fig.11 Application diagram.

(1) Omit this capacitor when a microcontroller is used.

17 PACKAGE OUTLINE

QFP44: plastic quad flat package; 44 leads (lead length 1.3 mm); body 10 x 10 x 1.75 mm **SOT307-2**



DIMENSIONS (mm are the original dimensions)

UNIT	A max.	A_1	A_2	A_3	b_p	c	$D^{(1)}$	$E^{(1)}$	e	H_D	H_E	L	L_p	v	w	y	$Z_D^{(1)}$	$Z_E^{(1)}$	θ
mm	2.10	0.25 0.05	1.85 1.65	0.25	0.40 0.20	0.25 0.14	10.1 9.9	10.1 9.9	0.8	12.9 12.3	12.9 12.3	1.3	0.95 0.55	0.15	0.15	0.1	1.2 0.8	1.2 0.8	10^o 0^o

Note

1. Plastic or metal protrusions of 0.25 mm maximum per side are not included.

OUTLINE VERSION	REFERENCES				EUROPEAN PROJECTION	ISSUE DATE
	IEC	JEDEC	EIAJ			
SOT307-2						~~95-02-04~~ 97-08-01

18 SOLDERING

18.1 Introduction to soldering surface mount packages

This text gives a very brief insight to a complex technology. A more in-depth account of soldering ICs can be found in our *"Data Handbook IC26; Integrated Circuit Packages"* (document order number 9398 652 90011).

There is no soldering method that is ideal for all surface mount IC packages. Wave soldering can still be used for certain surface mount ICs, but it is not suitable for fine pitch SMDs. In these situations reflow soldering is recommended.

18.2 Reflow soldering

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 methods exist for reflowing; for example, convection or convection/infrared heating in a conveyor type oven. Throughput times (preheating, soldering and cooling) vary between 100 and 200 seconds depending on heating method.

Typical reflow peak temperatures range from 215 to 250 °C. The top-surface temperature of the packages should preferable be kept below 220 °C for thick/large packages, and below 235 °C for small/thin packages.

18.3 Wave soldering

Conventional single wave soldering is not recommended for surface mount devices (SMDs) or printed-circuit boards with a high component density, as solder bridging and non-wetting can present major problems.

To overcome these problems the double-wave soldering method was specifically developed.

If wave soldering is used the following conditions must be observed for optimal results:

- Use a double-wave soldering method comprising a turbulent wave with high upward pressure followed by a smooth laminar wave.
- For packages with leads on two sides and a pitch (e):
  - larger than or equal to 1.27 mm, the footprint longitudinal axis is **preferred** to be parallel to the transport direction of the printed-circuit board;
  - smaller than 1.27 mm, the footprint longitudinal axis **must** be parallel to the transport direction of the printed-circuit board.

  The footprint must incorporate solder thieves at the downstream end.
- For packages with leads on four sides, the footprint must be placed at a 45° angle to the transport direction of the printed-circuit board. The footprint must incorporate solder thieves downstream and at the side corners.

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.

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.

18.4 Manual soldering

Fix the component by first soldering two diagonally-opposite end leads. Use a low voltage (24 V or less) soldering iron 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.

18.5 Suitability of surface mount IC packages for wave and reflow soldering methods

PACKAGE	SOLDERING METHOD	
	WAVE	REFLOW[1]
BGA, LFBGA, SQFP, TFBGA	not suitable	suitable
HBCC, HLQFP, HSQFP, HSOP, HTQFP, HTSSOP, SMS	not suitable[2]	suitable
PLCC[3], SO, SOJ	suitable	suitable
LQFP, QFP, TQFP	not recommended[3][4]	suitable
SSOP, TSSOP, VSO	not recommended[5]	suitable

Notes

1. All surface mount (SMD) packages are moisture sensitive. Depending upon the moisture content, the maximum temperature (with respect to time) and body size of the package, there is a risk that internal or external package cracks may occur due to vaporization of the moisture in them (the so called popcorn effect). For details, refer to the Drypack information in the *"Data Handbook IC26; Integrated Circuit Packages; Section: Packing Methods"*.
2. These packages are not suitable for wave soldering as a solder joint between the printed-circuit board and heatsink (at bottom version) can not be achieved, and as solder may stick to the heatsink (on top version).
3. If wave soldering is considered, then the package must be placed at a 45° angle to the solder wave direction. The package footprint must incorporate solder thieves downstream and at the side corners.
4. Wave soldering is only suitable for LQFP, TQFP and QFP packages with a pitch (e) equal to or larger than 0.8 mm; it is definitely not suitable for packages with a pitch (e) equal to or smaller than 0.65 mm.
5. Wave soldering is only suitable for SSOP and TSSOP packages with a pitch (e) equal to or larger than 0.65 mm; it is definitely not suitable for packages with a pitch (e) equal to or smaller than 0.5 mm.

19 DATA SHEET STATUS

DATA SHEET STATUS[1]	PRODUCT STATUS[2]	DEFINITIONS
Objective data	Development	This data sheet contains data from the objective specification for product development. Philips Semiconductors reserves the right to change the specification in any manner without notice.
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SCA 72

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