



## Digital Audio Decoder/Processor for A2 and NICAM Television/Video Recorders

DATASHEET

### Key Features

#### ■ Full-Automatic Multi-Standard Demodulation

- B / G / I / L / M / N / D / K Standards
- Mono AM and FM
- FM 2-Carrier (German and Korean Zweiton) and NICAM

#### ■ Multi-Channel Capability

- 3 I<sup>2</sup>S digital inputs, S/PDIF (pass-thru/out)
- 1 I<sup>2</sup>S digital output (shared with one of the I<sup>2</sup>S digital inputs)
- 5.1 analog outputs
- Dolby® Pro Logic®
- Dolby® Pro Logic II®

#### ■ Sound Processing: Loudspeaker

- ST royalty-free processing: ST WideSurround, ST OmniSurround (Virtual Dolby® Surround and Virtual Dolby® Digital compliant) and ST Dynamic Bass
- SRS® WOW™, SRS® TruSurround XT™ (Virtual Dolby® Surround and Virtual Dolby® Digital compliant)
- Independent Volume / Balance
- Smart Volume Control (SVC), 5-band equalizer and loudness

#### ■ Sound Processing: Headphone

- Smart Volume Control (SVC), Bass-Treble, Loudness and SRS® TruBass™
- Independent Volume / Balance

#### ■ Analog Audio Matrix

- 4 stereo inputs
- 3 stereo outputs
- THRU mode
- 2 V<sub>RMS</sub> capability

#### ■ Audio Delay for Audio Video Synchronization

- Embedded stereo delay up to **90 ms** when processing at 32KHz (demodulator input mode) and up to **60 ms** when processing at 48KHz (SCART only input mode)
- Independent delay on headphone and loudspeaker channels

The STV82x7 family, based on audio digital signal processors (DSP), performs high quality and advanced dedicated digital audio processing. These devices provide all of the necessary resources for automatic detection and demodulation of analog audio transmissions for European and Asian terrestrial TV broadcasts.

Virtual or true, multi-channel capabilities and easy digital links make them ideal for digital audio low cost consumer applications. Starting from enhanced stereo up to independent control of 5 loudspeakers and a subwoofer (5.1 channels), the STV82x7 family offers standard and advanced features plus sound enhancements, spatial and virtual effects to enhance television viewer comfort and entertainment.

### Typical Applications

- Analog and digital TV with virtual surround sound
- Analog and digital TV with multi-channel surround sound
- DVD and HDD recorders
- "Palm size" portable TV

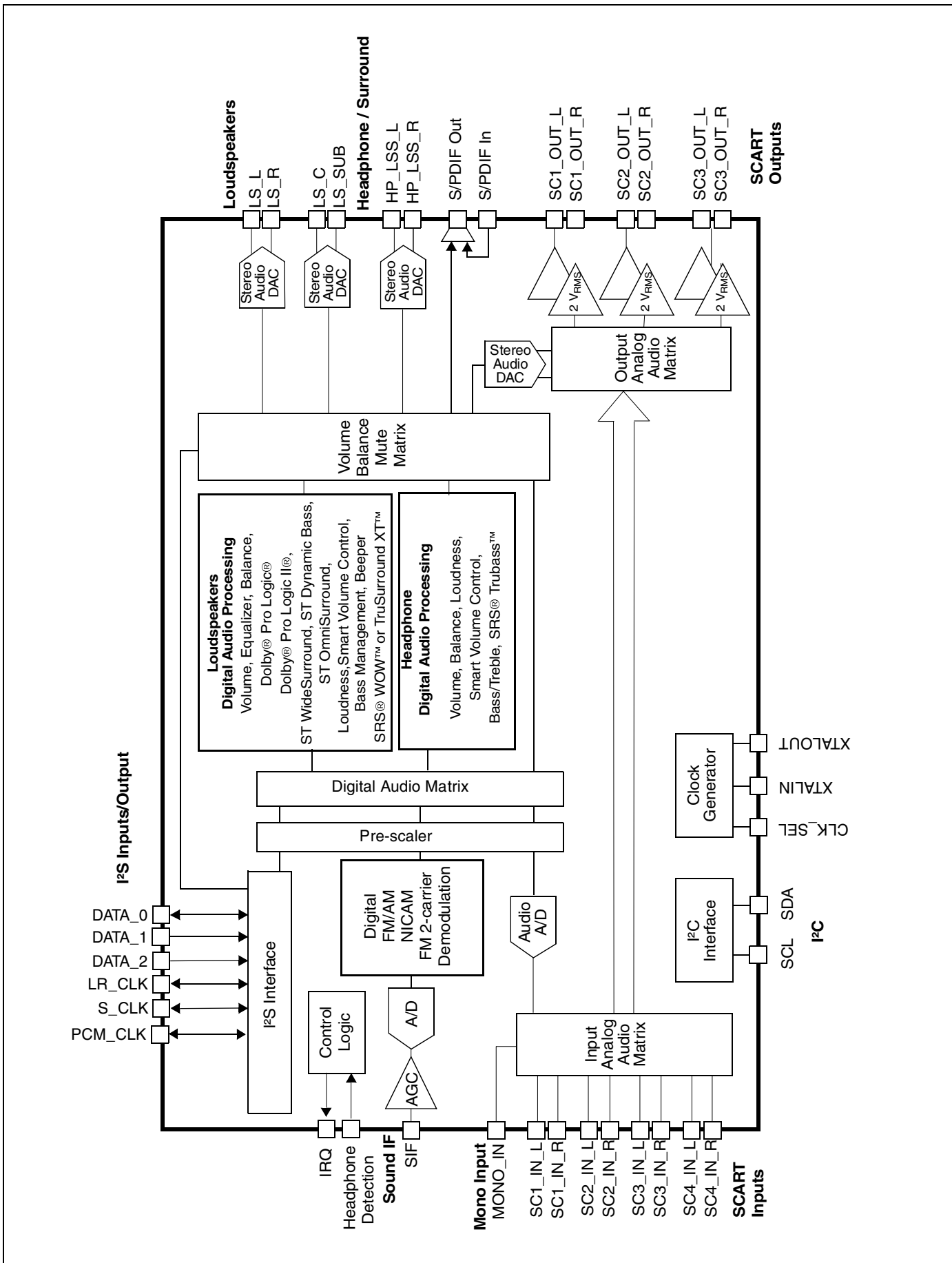


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# Block Diagram



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# 1 General Description

The STV82x7 is a multistandard TV sound demodulator and audio processor which integrates **SRS® WOW™**, **SRS® TruSurround XT™**, **Dolby® Pro Logic®**, **Dolby® Pro Logic II®**, **Virtual Dolby® Surround (VDS)** and **Virtual Dolby® Digital (VDD)** capability.

ST advanced algorithms such as **ST OmniSurround**, **ST WideSurround**, **ST Dynamic Bass** are also available in this audio sound processor. **ST OmniSurround** is a certified **Dolby®** algorithm for the **Virtual Dolby® Digital (VDD)** and the **Virtual Dolby® Surround (VDS)**. When using VDD or VDS, either a **Dolby® Digital** or a **Pro Logic®** (or **Pro Logic II®**) decoder is mandatory respectively.

This chip performs **automatic multistandard analog TV stereo sound identification and demodulation** (no specific I<sup>2</sup>C programming is required). It offers various audio processing functions such as equalization, loudness, beeper, volume, balance, and surround effects. It provides a cost-effective solution for analog and digital TV designs.

The STV82x7 is perfectly suited to current and future digital TV platforms, based on audio/video digital chips (**STD2000**, **DTV100 platform**) which include an internal digital decoder (**MPEG**, **Dolby® Digital...**). In the case where a Dolby® Digital decoder is embedded in the audio/video digital chip, Virtual Dolby® Digital can be obtained.


For the **CTV100/120** platforms, this device is offered as an alternative solution to the first-generation chassis that uses the STV82x6.

Table 1: STV82x7 Version List

	STV8217	STV8237	STV8247		STV8257			STV8267		STV8277		STV8287	
			STV8247D	STV8247DSX	STV8257D	STV8257DSX	STV8257SX	STV8267D	STV8267DSX	STV8277D	STV8277DSX	STV8287D	STV8287DSX
<b>Demodulation</b>													
FM 2 Carrier and NICAM	X	X	X	X	X	X	X	X	X	X	X	X	X
<b>Multi-Channel Capabilities</b>													
Analog loudspeakers output number	2.1	2.1	2.1	2.1	2.1	2.1	2.1	5.1	5.1	5.1	5.1	5.1	5.1
I <sup>2</sup> S In (exclusive with I <sup>2</sup> S Out)	1	1	1	1	3	3	3	1	1	3	3	3	3
S/PDIF (Pass-thru or Output)	1	1	1	1	1	1	1	1	1	1	1	1	1
Virtual Dolby® Surround			X	X	X	X		X	X	X	X	VDS PLII	VDS PLII
Virtual Dolby® Digital capability <sup>1</sup>					X	X	X			X	X	X	X
Dolby® Pro Logic® (DPLI) or Dolby® Pro Logic II® (DPLII)			DPLI (internal)	DPLI (internal)	DPLI (internal)	DPLI (internal)		DPLI	DPLI	DPLI	DPLI	DPLII	DPLII
<b>Audio Processing</b>													
SRS® WOW™ (WOW)		X											
SRS® TruSurround XT™				X		X	X		X		X		X
ST Voice, ST Dynamic Bass	X	X	X	X	X	X	X	X	X	X	X	X	X
ST WideSurround, ST OmniSurround <sup>2</sup>	X	X	X	X	X	X	X	X	X	X	X	X	X

1. **Dolby® Digital** Bypass capability or **Virtual Dolby® Digital** are obtained with the use of an external **Dolby® Digital** decoder (for example STD2000).
2. When using **Virtual Dolby® Digital** or **Virtual Dolby® Surround** with **ST OmniSurround** or **SRS® TruSurround XT™** a **Dolby® Digital** or a **Pro Logic®** (or **Pro Logic II®**) decoder is mandatory.

Figure 1: Package Ordering Information

<p><b>Order Code:</b>                  STV82x7 (Tray)                  STV82x7/T (Tape &amp; Reel)</p>	
<p><b>For Example:</b> STV8257DSX/T will be delivered in Tape &amp; Reel conditioning</p>	



## 1.1 STV82x7 Overview

### 1.1.1 Core Features

- Single audio source processing:
  - IF source and/or analog stereo input (SCART)
  - one digital source with a maximum of 6 synchronous channels (5.1 is obtained across three I<sup>2</sup>S)
- SIF input signal with Automatic Gain Control (AGC)
- Digital Demodulator with automatic standard detection and demodulation for AM, FM mono, FM 2 carriers (German or Korean FM 2-carrier) and NICAM
- Audio processor working at 32 kHz, 44.1 kHz or 48 kHz with specific features:
  - For Loudspeakers (L, R, L<sub>S</sub>, R<sub>S</sub>, SubW, C):
    - Dolby® Pro Logic II ® Decoder with Bass Management
    - SRS® WOW™ or TruSurround XT™ including Virtual Dolby® Surround and Virtual Dolby® Digital
    - ST WideSurround
    - ST OmniSurround
    - ST Dynamic Bass
    - 5-band Equalizer or Bass-Treble
    - Loudness
    - Smart Volume Control
    - Volume/Balance/Soft-mute
    - Beeper
    - Video Processing Delay Compensation
  - For Headphone:
    - SRS® TruBass™
    - Smart Volume Control
    - Bass-Treble
    - Loudness
    - Volume/Balance/Soft-mute
    - Beeper
    - Video Processing Delay Compensation
- Shared outputs for headphone and loudspeakers surround channels:
- Analog matrix with:
  - five external inputs:
    - four SCART inputs (2 V<sub>RMS</sub> capable)
    - one analog mono input (0.5 V<sub>RMS</sub>)
  - one internal input from a digital matrix via a DAC
  - three external outputs (2 V<sub>RMS</sub> capable)
  - one internal output for the digital matrix (using an internal ADC)
- Digital matrix with:
  - three input modes (Demodulator/SCART, SCART only and I<sup>2</sup>S)
  - three stereo outputs (Loudspeakers, Headphone and SCART)
- High-end audio DAC
- S/PDIF pass-thru/output for connection with an external amplifier/decoder
- Internal multiplexer for the S/PDIF output (to share the internal S/PDIF output and the S/PDIF output generated by the external decoder of the digital broadcast)

- Specific stand-by mode (Loop-through)
- Control by I<sup>2</sup>C bus (two I<sup>2</sup>C addresses)
- System PLL and Clock Generation using either a single quartz oscillator or a differential clock input

### 1.1.2 Software Information

The different software combinations are listed in [Table 2](#).

**Table 2: Input/Output Software Configurations**

Input (Number of Channels)	Output (Number of Channels)		
	2 (+1)	4 (+1)	5 (+1)
1	ST WideSurround or SRS® WOW™		
2 (L and R)	ST WideSurround or SRS® WOW™		
2 (L <sub>T</sub> and R <sub>T</sub> )	ST WideSurround or SRS® TruSurround XT™ or ST OmniSurround or Dolby® Pro Logic® + SRS® TruSurround XT™ or Dolby® Pro Logic® + ST OmniSurround	Dolby® Pro Logic®	
4 (+1)	SRS® TruSurround XT™ or ST OmniSurround or Downmix	No processing	
5 (+1)	SRS® TruSurround XT™ or ST OmniSurround or Downmix	Downmix	No processing

*Note: 1 In addition to the above sound processing, it is always possible to add ST Voice and also ST Dynamic Bass algorithms.*

*2 The SRS® TruSurround® and ST OmniSurround are approved by Dolby as Virtual Dolby Surround (VDS) and Virtual Dolby Digital (VDD).*

*The SRS® TruSurround XT™ system is composed of:*

- SRS® TruSurround®
- SRS® WOW™

*The SRS® WOW™ system includes:*

- SRS® 3D Mono/Stereo™
- SRS® Dialog Clarity™
- SRS® TruBass™

### 1.1.3 Device Input Modes

- Demodulator only mode (with output  $f_S = 32$  kHz)
- Demodulator and SCART mode (with output  $f_S = 32$  kHz)
- SCART only mode (with output  $f_S = 48$  kHz)

- I<sup>2</sup>S mode (with output  $f_s = 32, 44.1$  or  $48$  kHz)
  - External audio input interface using 3 x I<sup>2</sup>S (for decoded streams such as Dolby® Digital and/or standard stereo streams)

#### 1.1.4 Electrical Features

Multi Power Supply: 1.8 V, 3.3 V and 8 V.

Power Consumption:

- lower than 1 W in Functional mode (full features)
- 200 mW in Loop-through mode corresponding to Switch-off of all digital blocks

## 1.2 Typical Applications

The STV82x7 is specified to enable flexible, analog and digital TV chassis design (refer to [Figure 2](#), [Figure 3](#), [Figure 4](#) and [Figure 5](#)).

The main considerations are:

- all necessary connections between devices can be provided through the TV set,
- pseudo stand-by mode used to copy to VCR or the DVD sources when the TV set is OFF,
- possible application compatibility with STV82x6 (TQFP80 package) TV design,
- pin-to-pin compatibility with STV82x8 (TQFP80 package) TV design.

The STV82x7 is used to process a single audio source (analog or digital). However, it is possible to process two audio sources simultaneously using an STV82x7 interconnection (two chips can be easily connected).

In the case of a single audio source, it is possible to hear and record in the same time: the same audio stream can be simultaneously output on headphone, loudspeakers, S/PDIF and the SCART connectors.

*Note: Headphone and loudspeakers can be used simultaneously for dual-language purposes or for different sound settings (e.g. volume). In this case, certain restrictions occur (see [Section 4.2: Audio Processing](#)).*

For more connections, the SCART-to-SCART path can be used. The use of these full analog paths implies that the sound is not digitally processed.

Figure 2: STV8237 Typical Application (Enhanced Stereo)

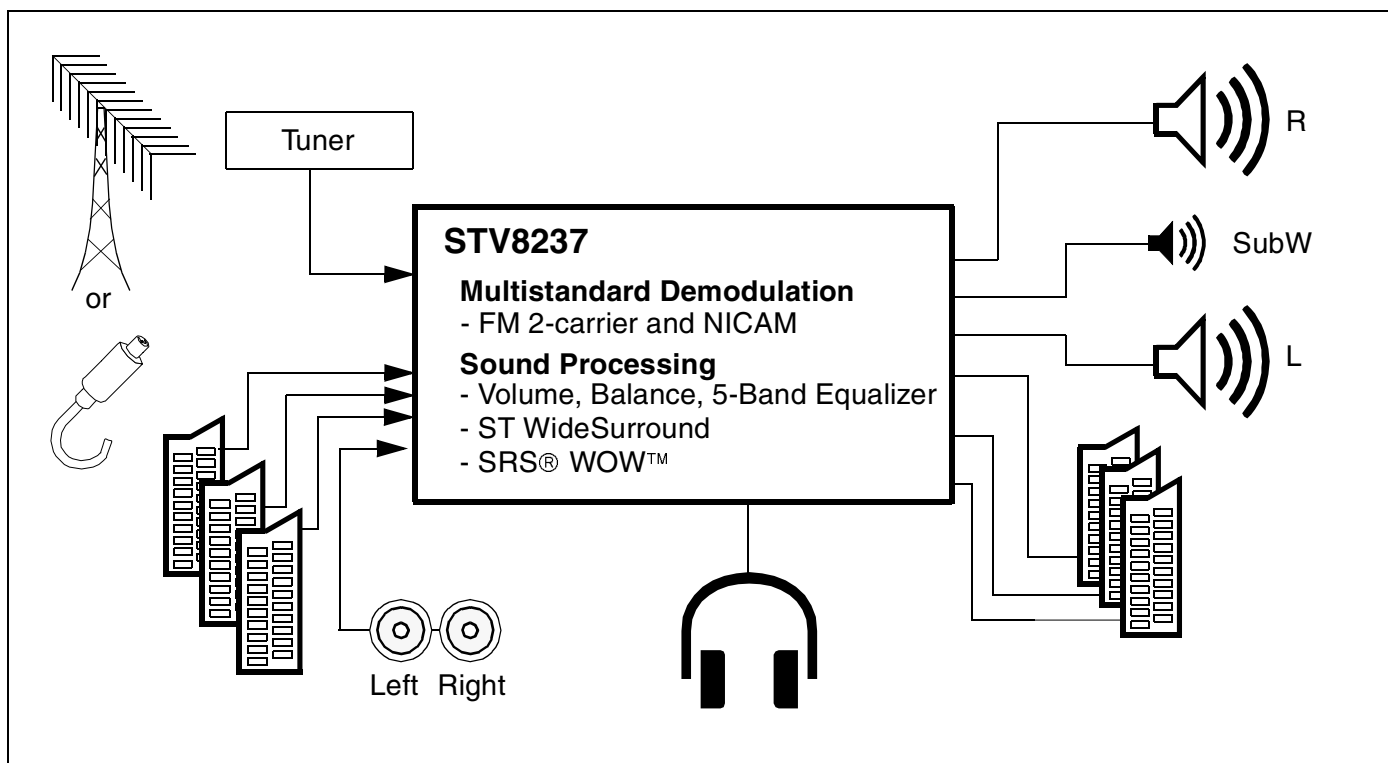


Figure 3: STV8247 Typical Application (Analog Virtual Sound)

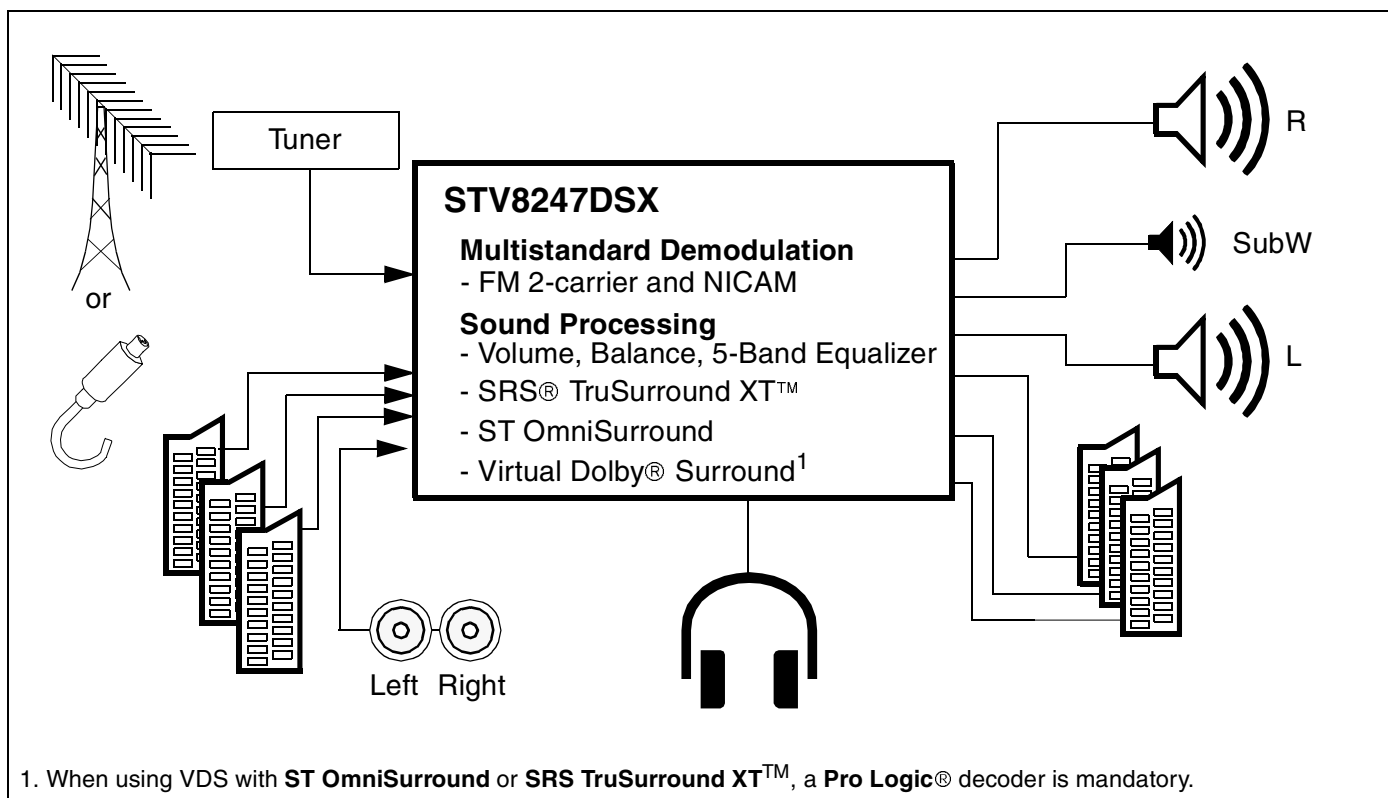


Figure 4: STV8257 Typical Application (Digital: Virtual Sound)

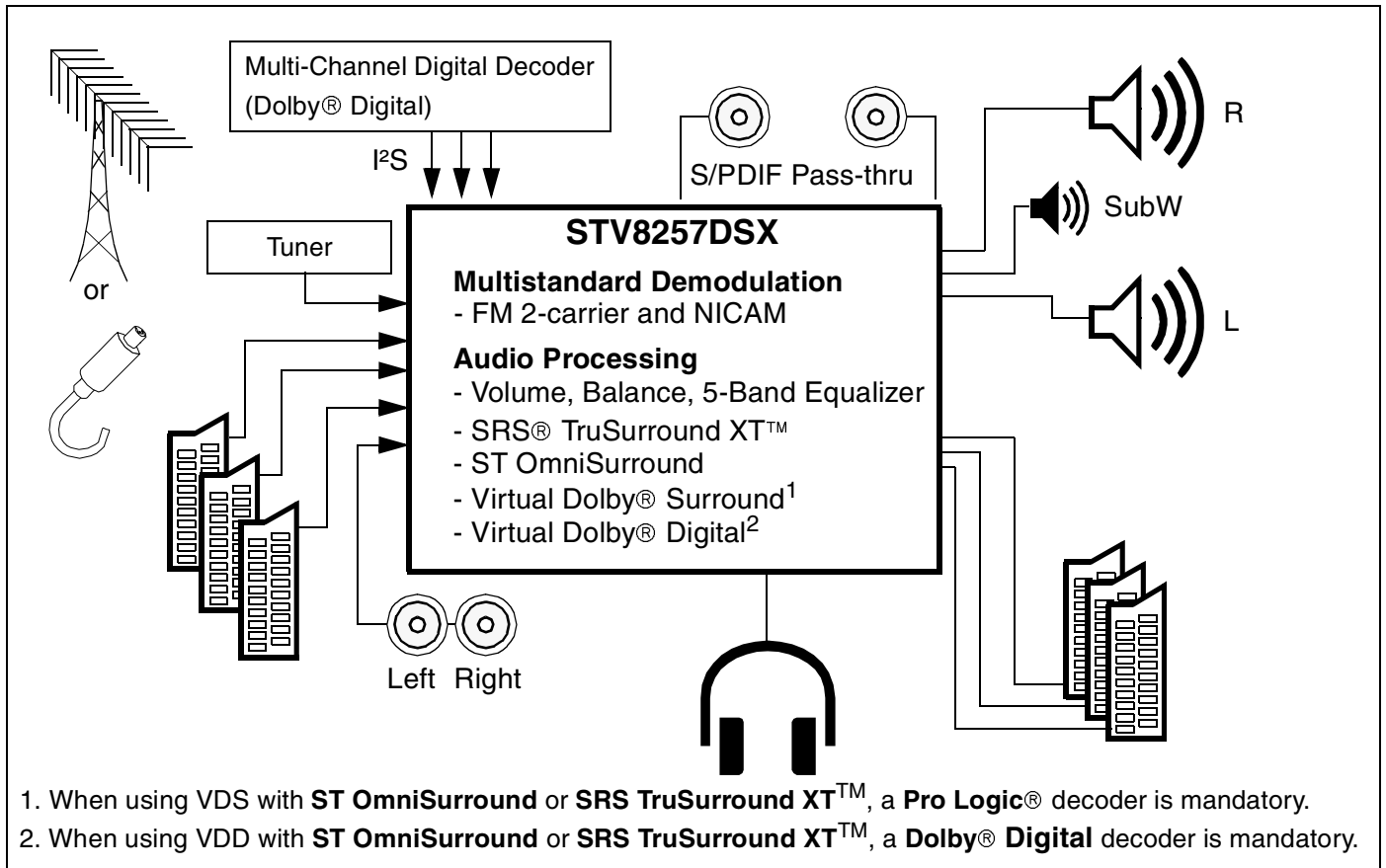


Figure 5: STV8277 Typical Application (Digital TV: Multi-Channel and Virtual Sound)

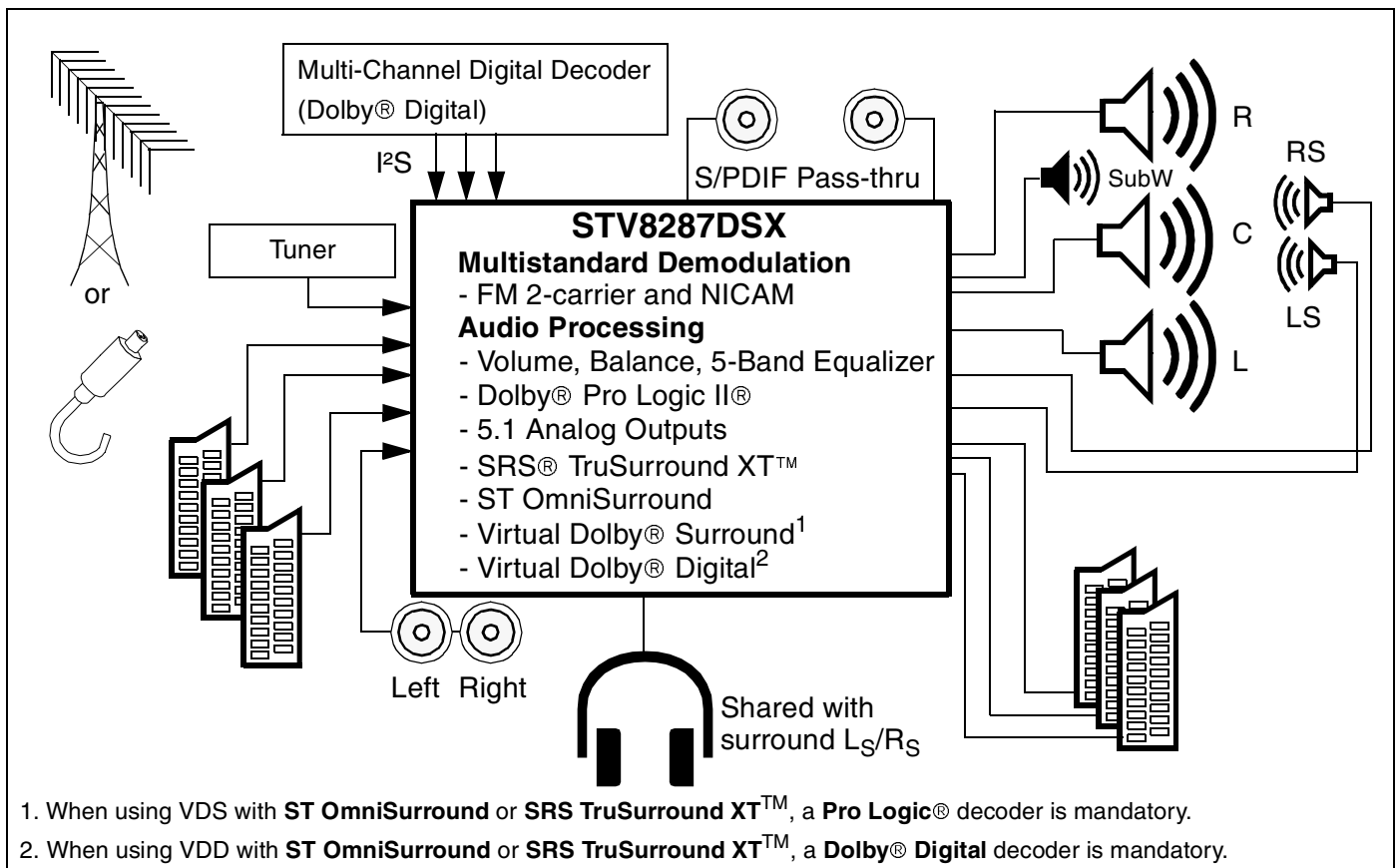
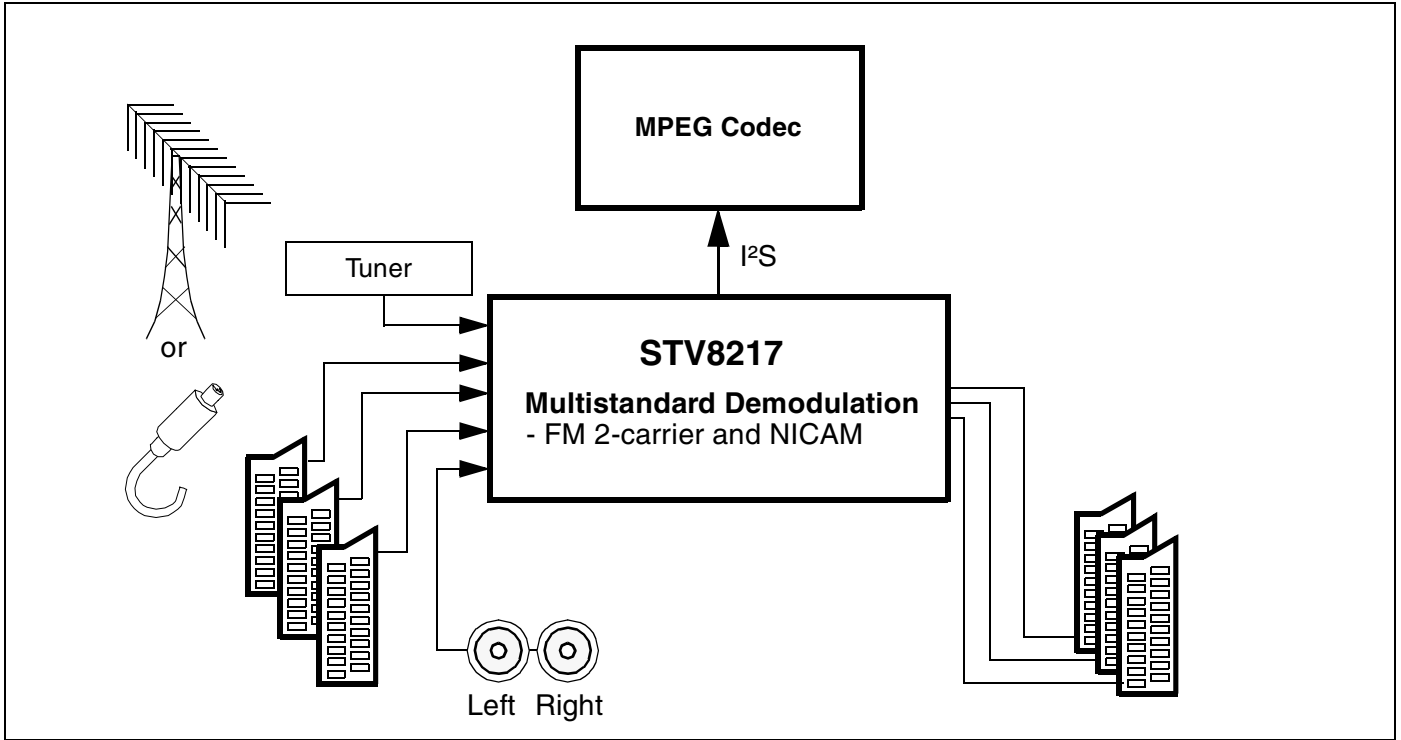


Figure 6: STV8217 Typical Application (Digital Recorder)



### 1.3 Pin Descriptions and Application Diagrams

- AP = Analog Power
- DP = Digital Power
- I = Input
- O = Output
- OD = Open-Drain
- B = Bi-Directional
- A = Analog

**Table 3: TQFP80 Pin Description (Sheet 1 of 3)**

Pin No.	STV82x7 Pin Name	Type (STV82x7)	Function for STV82x7 (Function for STV82x6 in italic characters)	STV82x6 Pin Name
1	SC1_OUT_L	A	SCART1 Audio Output Left	AO1L
2	SC1_OUT_R	A	SCART1 Audio Output Right	AO1R
3	VCC_H	AP	8 V Power for Audio I/O & ESD	<i>Not connected</i>
4	GND_H	AP	High Current Ground for Audio Outputs	<i>Connected to Ground</i>
5	SC3_OUT_L	A	SCART3 Audio Output Left	<i>Not connected</i>
6	SC3_OUT_R	A	SCART3 Audio Output Right	<i>Not connected</i>
7	VCC33_SC	AP	3.3 V Power for Audio Buffers & DAC / ADC	VDDC
8	GND33_SC	AP	Ground for Audio Buffers & DAC / ADC	GNDC
9	SC1_IN_L	A	SCART1 Audio Input Left	AI1L
10	SC1_IN_R	A	SCART1 Audio Input Right	AI1R
11	VREFA	A	Audio Bias Voltage Decoupling 1.55 V (Switched $V_{REF}$ decoupling pin for Audio Converters (VMCP))	VMC1
12	GND_SA	AP	Ground for DACs	<i>Connected to Ground</i>
13	VBG	A	Bandgap Voltage Reference Decoupling 1.2 V ( $V_{REF}$ decoupling pin for Audio Converters (VMC))	VMC2
14	SC2_IN_L	A	SCART2 Audio Input Left	AI2L
15	SC2_IN_R	A	SCART2 Audio Input Right	AI2R
16	VCC33_LS	AP	3.3 V Power for Audio DACs (3.3 V Power Supply for Audio Buffers and SCART)	VDDA
17	GND33_LS	AP	Ground for Audio DACs (Ground for Audio Buffers and SCART)	GNDAH
18	SC2_OUT_L	A	SCART2 Audio Output Left	AO2L
19	SC2_OUT_R	A	SCART2 Audio Output Right	AO2R
20	VCC_NISO	AP	Polarization of the NISO (connected to 3.3 V) (8 V / 5 V Power supply for SCART & Audio buffers)	VDDH
21	VSS33_CONV	AP	Ground for DAC 1.8 to 3.3 V Converters	<i>Connected to Ground</i>
22	VDD33_CONV	AP	3.3 V Power for DAC 1.8 to 3.3 V Converters (Voltage Reference for Audio buffers)	VREFA

Table 3: TQFP80 Pin Description (Sheet 2 of 3)

Pin No.	STV82x7 Pin Name	Type (STV82x7)	Function for STV82x7 (Function for STV82x6 in <i>italic characters</i> )	STV82x6 Pin Name
23	SC3_IN_L	A	SCART3 Audio Input Left	AI3L
24	SC3_IN_R	A	SCART3 Audio Input Right	AI3R
25	SCL_FLT	A	SCART Filtering Left	<i>Not connected</i>
26	SCR_FLT	A	SCART Filtering Right ( <i>Bandgap Voltage Source Decoupling</i> )	BGAP
27	LS_C	A	Center Output	<i>Not connected</i>
28	LS_L	A	Left Loudspeaker Output	LSL
29	LS_R	A	Right Loudspeaker Output	LSR
30	LS_SUB	A	Subwoofer Output	SW
31	HP_LSS_L	A	Left Headphone Output or Left Surround Output	HPL
32	HP_LSS_R	A	Right Headphone Output or Right Surround Output	HPR
33	VSS18_CONV	DP	Ground for Digital part of the DAC/ADC ( <i>Substrate Analog/Digital Shield</i> )	GNSA
34	VDD18_CONV	DP	1.8 V Power for Digital part of the DAC/ADC	<i>Not connected</i>
35	HP_DE $\bar{T}$	I	Headphone Detection	HP $\bar{D}$
36	ADR_SEL	I	Hardware Address selection for I <sup>2</sup> C Bus	ADR
37	VSS18	DP	Ground for Digital part	<i>Connected to Ground</i>
38	VDD18	DP	1.8 V Power for Digital part	<i>Not connected</i>
39	SCL	OD	I <sup>2</sup> C Clock Input	SCL
40	SDA	OD	I <sup>2</sup> C Data I/O	SDA
41	VSS18	DP	Ground for Digital part	<i>Connected to Ground</i>
42	VDD18	DP	1.8 V Power for Digital part ( <i>5 V Power Regulator Control</i> )	REG
43	R $\bar{S}T$	I	Main Reset Input	R $\bar{E}S\bar{E}T$
44	S/PDIF_IN	I	Serial Audio Data Input ( <i>System Clock output</i> )	SY $\bar{S}C\bar{K}$
45	S/PDIF_OUT	O	Serial Audio Data Output ( <i>I<sup>2</sup>S Master Clock output</i> )	MCK
46	VDD33_IO1	DP	3.3 V Power for Digital part	VDD1
47	VSS33_IO1	DP	Ground for Digital part	GND1
48	CK_TST_CTRL	D	To be Grounded	<i>Not connected</i>
49	VSS18	DP	Ground for Digital part	GNDSP
50	VDD18	DP	1.8 V Power for Digital part	<i>Not connected</i>
51	CLK_SEL	I	Clock Input Format Selection	<i>Not connected</i>
52	XTALIN_CLKXTP	I	Crystal Oscillator Input or Differential Input Positive ( <i>Crystal Oscillator Input</i> )	XTI

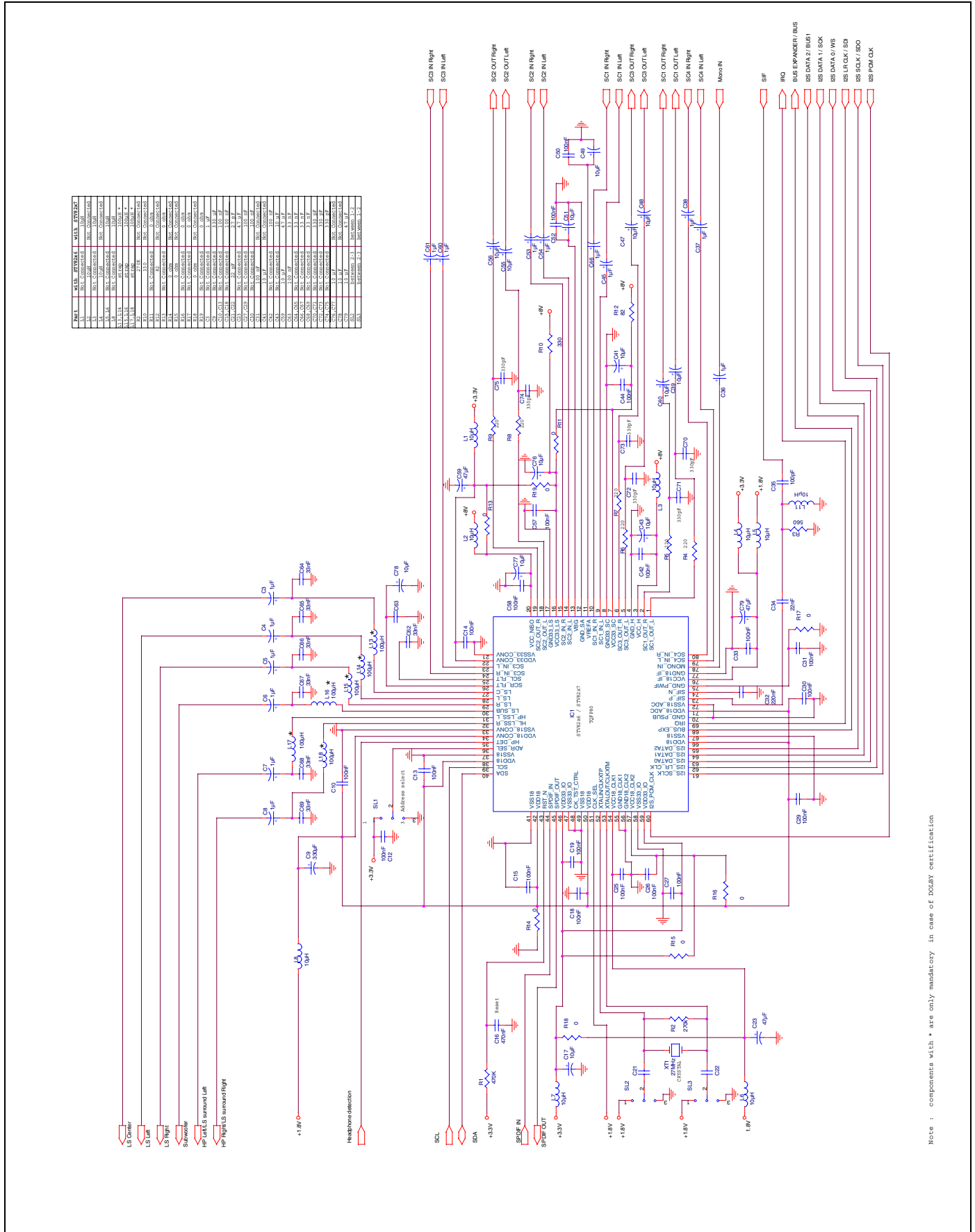


Table 3: TQFP80 Pin Description (Sheet 3 of 3)

Pin No.	STV82x7 Pin Name	Type (STV82x7)	Function for STV82x7 (Function for STV82x6 in italic characters)	STV82x6 Pin Name
53	XTALOUT_CLKXTM	O	Crystal Oscillator Output or Differential Input Negative ( <i>Crystal Oscillator Output</i> )	XTO
54	VCC18_CLK1	AP	1.8 V Power for Clock PLL Analog & Crystal Oscillator 1/2 ( <i>3.3 V Power supply for Analog PLL Clock</i> )	VDDP
55	GND18_CLK1	AP	Ground for Clock PLL Analog & Crystal Oscillator 1/2	GNDP
56	GND18_CLK2	DP	Ground for Clock PLL Digital 1/2	GND2
57	VCC18_CLK2	DP	1.8 V Power for Clock PLL Digital 1/2 ( <i>3.3 V Power supply for Digital core, DSPs &amp; IO Cells</i> )	VDD2
58	VSS33_IO2	DP	Ground for Digital IO pins 60 to 69	<i>Connected to Ground</i>
59	VDD33_IO2	DP	3.3 V power for Digital IO pins 60 to 69	<i>Not connected</i>
60	I2S_PCM_CLK	I/O	I <sup>2</sup> S Slave Clock Input/Output Channel 1, 2 & 3	<i>Not connected</i>
61	I2S_SCLK	I/O	I <sup>2</sup> S Clock Input/Output Channel 1, 2 & 3 (I <sup>2</sup> S bus data output)	SDO
62	I2S_LR_CLK	I/O	I <sup>2</sup> S Word Select Input/Output Channel 1,2 & 3 ( <i>Stereo Detection output / I<sup>2</sup>S Bus Data input</i> )	ST/SDI
63	I2S_DATA0	I/O	I <sup>2</sup> S Data Input/Output Stereo Channel 1 ( <i>I<sup>2</sup>S Bus Word Select output</i> )	WS
64	I2S_DATA1	I	I <sup>2</sup> S Data Input Stereo Channel 2 ( <i>I<sup>2</sup>S Bus Clock output</i> )	SCK
65	I2S_DATA2	I	I <sup>2</sup> S Data Input Stereo Channel 3 ( <i>Bus Expander Output 1</i> )	BUS1
66	VDD18	DP	1.8 V Power for Digital Core & I/O Cells Pin	<i>Not connected</i>
67	VSS18	DP	Ground for Digital Core & I/O Cells Pin	<i>Connected to Ground</i>
68	BUS_EXP	O	Bus Expander Function ( <i>Bus Expander Output 2</i> )	BUS0
69	IRQ	O	Interrupt Request to Microprocessor	IRQ
70	GND_PSUB	AP	Ground Substrate Connection	<i>Connected to Ground</i>
71	VDD18_ADC	DP	VDD 1.8 V for ADC (Digital Part)	<i>Not connected</i>
72	VSS18_ADC	DP	Ground to Complement 1.8 V VDD for ADC	<i>Connected to Ground</i>
73	SIF_P	A	Sound IF input (positive)	SIF
74	SIF_N	A	Sound IF input (negative) (ADC V <sub>TOP</sub> Decoupling pin)	VTOP
75	GNDPW_IF	AP	Polarization for the IF block ( <i>Voltage Reference for AGC Decoupling pin</i> )	VREFIF
76	VCC18_IF	AP	1.8 V Power for IF AGC & ADC	VDDIF
77	GND18_IF	AP	Ground for IF AGC & ADC	GNDIF
78	MONO_IN	A	Mono Input (for AM Mono)	MONOIN
79	SC4_IN_L	A	SCART4 Audio Input Left	<i>Not connected</i>
80	SC4_IN_R	A	SCART4 Audio Input Right	<i>Not connected</i>



Figure 8: STV82x6/STV82x7 Compatible Application Electrical Diagram



Note : components with \* are only mandatory in case of DO28Y certification

Figure 9: STV82x7/STV82x8 Compatible Application Electrical Diagram (TQFP80)

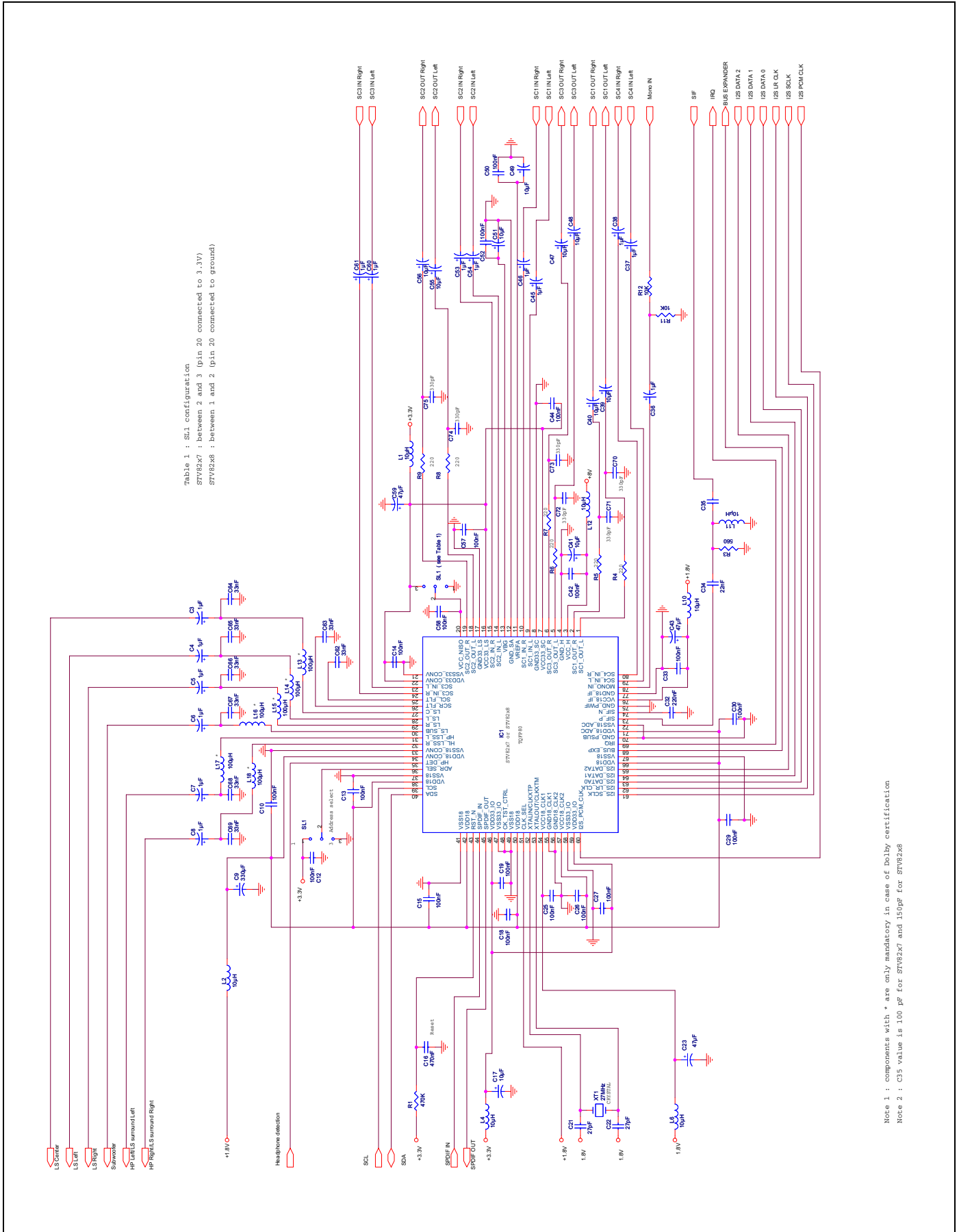


Table 1 : SCL configuration  
STV82x7 : between 2 and 3 (pin 20 connected to 3.3V)  
STV82x8 : between 1 and 2 (pin 20 connected to ground)

Note 1 : components with \* are only mandatory in case of Dolby certification  
Note 2 : C35 value is 100 pF for STV82x7 and 150pF for STV82x8



## 2 System Clock

The System Clock integrates 2 independent frequency synthesizers.

The first frequency synthesizer can be used in one of two modes:

- In Mode 1, it is used by the demodulator, and the frequency is 49.152 MHz.
- In Mode 2, it is used by the I<sup>2</sup>S input and is synchronous with the input frequency ( $f_S = 32, 44.1$  or 48 kHz) and the frequency is 49.152 MHz (for  $f_S = 32$  or 48 kHz) or 45.1584 MHz (for  $f_S = 44.1$  kHz).

The second frequency synthesizer is used by the DSP core and can be adjusted between 100 and 150 MHz depending on the application (around 106 MHz at reset value).

In I<sup>2</sup>S output mode, clocks are generated by synthesizer 1.

The default values are designed for a **standard 27 MHz reference frequency** provided by a stable single crystal or an external differential clock signal (for example, from the STV35x0) depending on the CLK\_SEL pin configuration (CLK\_SEL = 1 means a single crystal, 0 means an external differential clock). The 27 MHz value is the recommended frequency for minimizing potential RF interference in the application. The sinusoidal clock frequency, and any harmonic products, remain outside the TV picture and sound IFs (PIF/SIF) and Band-I RF.

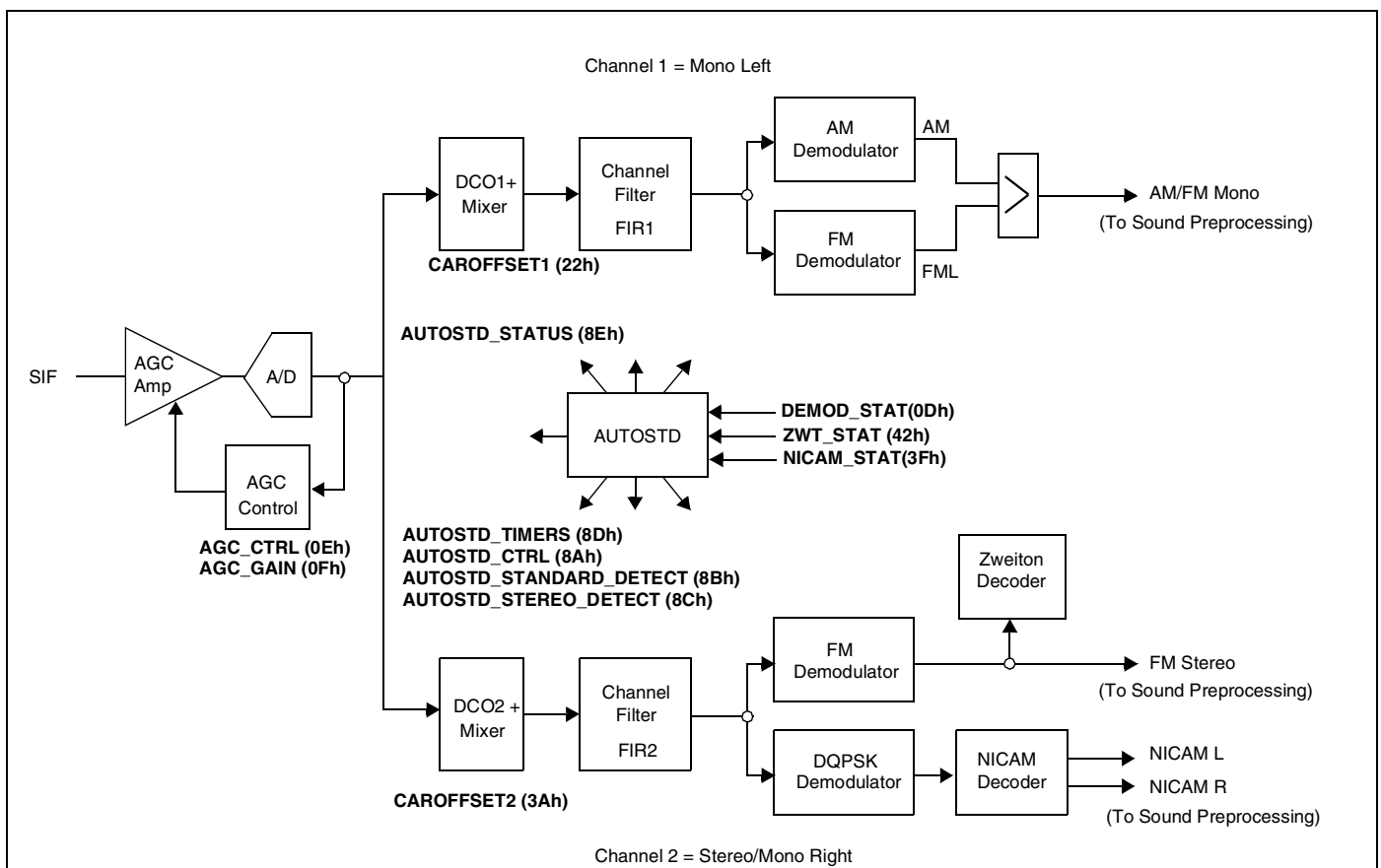
*Note: A change in the reference frequency is compatible with other default I<sup>2</sup>C programming values, including those of the built-in Automatic Standard Recognition System.*

### 3 Digital Demodulator

The Digital Demodulator (see [Figure 10](#)) is composed of two channels. The first channel demodulates an FM or an AM signal. The second channel demodulates FM 2-carrier or NICAM signals (stereo demodulation).

All channel parameters are programmed automatically by the **built-in Automatic Standard Recognition System** (Autostandard) in order to find the correct sound standard. Channels can also be programmed manually via the I<sup>2</sup>C interface for very specific standards not included among the known standards.

**Figure 10: Demodulator Block Diagram**



#### 3.1 Sound IF Signal

The Analog Sound Carrier IF is connected to the STV82x7 via the SIF pin. Before Analog-to-Digital Conversion (ADC), an Automatic Gain Control (AGC) is performed to adjust the incoming IF signal to the full scale of the ADC. A preliminary video rejection is recommended to optimize conversion and demodulation performances. The AGC system provides a gain value allowing for a wide range of SIF input levels and is activated for all standards, except L/L'. In this particular case, the sound carrier is AM-modulated and an automatic level adjustment would only damage the transmitted audio signal. A preset I<sup>2</sup>C parameter is provided to define the gain of the AGC used in Manual mode (Registers [AGC\\_CTRL](#) and [AGC\\_GAIN](#)).

*Note:* For optimum AM demodulation performance, it is recommended to use the MONO Input.

### 3.2 Demodulation

The demodulation system operates by default in Automatic mode. In this mode, the STV82x7 is able to **identify and demodulate any TV sound standard including NICAM and A2 systems** (see Table 4) without any external control via the I<sup>2</sup>C interface. It consists of the two demodulation channels (Channel 1 = Mono Left and Channel 2 = Mono Right/Stereo) to simultaneously process two sound carriers in order to handle all transmission modes (stereo and up to three mono languages). The **built-in Automatic Standard Recognition System** (Autostandard) automatically programs the appropriate bits in the I<sup>2</sup>C registers which are forced to Read-only mode for users (see Section 12.1). The programming is optimized for each standard to be identified and demodulated.

Each mono and stereo standard can be removed (or added) from the List of Standards to be recognized by programming registers [AUTOSTD\\_STANDARD\\_DETECT](#) and [AUTOSTD\\_STEREO\\_DETECT](#), respectively. The identified standard is displayed in register [AUTOSTD\\_STATUS](#) and any change to standard is flagged to the host system via pin IRQ. This flag must be reset by re-programming the MSBs of register [AUTOSTD\\_CTRL](#) while checking the detected standard status by reading registers [AUTOSTD\\_STATUS](#), [NICAM\\_STAT](#) and [ZWT\\_STAT](#). Moreover, the detection of Stereo mode during demodulation is also flagged in register [AUTOSTD\\_STATUS](#).

**Important:** L/L' and D/K standards cannot be automatically processed because the same frequency is used for the MONO carrier. An exclusive L/DK selection must be programmed in register [AUTOSTD\\_CTRL](#). This may be externally controlled by detecting the RF modulation sign, which is negative for all TV standards except L/L'.

To recover out-of standard FM deviations or the Sound Carrier Frequency Offset, additional I<sup>2</sup>C controls are provided without interfering with the Automatic Standard Recognition System (Autostandard).

**DK-NICAM Overmodulation Recovery:** Four different FM deviation ranges can be selected (via register [AUTOSTD\\_CTRL](#)) for the DK standard while the Autostandard system remains active. The maximum FM deviation is 500 kHz in DK Mono mode and 350 kHz in DK NICAM mode (limited by overlapping FM and NICAM spectrum values). The demodulated signal peak level (proportional to the FM deviation) is detected by the Peak Detector and written to registers [PEAK\\_DET\\_L](#) and [PEAK\\_DET\\_R](#). This value is used to implement Automatic Overmodulation Detection via an external I<sup>2</sup>C control.

**Important:** Only the selection of the 50 kHz FM deviation standard is compatible with the other DK-A2\* standards (DK1, DK2 or DK3). These standards must be removed from the list of standards (registers [AUTOSTD\\_STANDARD\\_DETECT](#) and [AUTOSTD\\_STEREO\\_DETECT](#)) when programming larger FM deviations reserved only for DK-NICAM standards.

Table 4: Recognized Standards

System	Sound Type	Type Name	Carrier 1 (MHz)	Carrier 2 (MHz)	FM Deviation			De-emphasis	Roll-off (%)	Pilot Frequency (kHz)
					Nom.	Max.	Over			
B/G	FM Mono		5.5							
	FM/NICAM		5.5	5.850	27	50	80	J17	40	
	FM 2-Carrier	A2	5.5	5.742	27	50	80	50 $\mu$ s		54.6875
D/K	FM Mono		6.5							
	FM/NICAM		6.5	5.850	27	50	80	J17	40	
D/K1	FM 2-Carrier	A2*	6.5	6.258				50 $\mu$ s		54.6875

Table 4: Recognized Standards (Continued)

System	Sound Type	Type Name	Carrier 1 (MHz)	Carrier 2 (MHz)	FM Deviation			De-emphasis	Roll-off (%)	Pilot Frequency (kHz)
					Nom.	Max.	Over			
D/K2	FM 2-Carrier	A2*	6.5	6.742				50 $\mu$ s		54.6875
D/K3	FM 2-Carrier	A2*	6.5	5.742				50 $\mu$ s		54.6875
I	FM Mono		6.0							
	FM/NICAM		6.0	6.552	27	50	80	J17	100	
L	AM/NICAM		6.5	5.850				J17	40	
M/N	FM Mono		4.5		15	27	50	75 $\mu$ s		
	FM 2-Carrier	A2+	4.5	4.724	15	27	50	75 $\mu$ s		55.069

For Chinese TV transmissions (DK-NICAM) which are subject to overmodulation, different FM deviations are proposed for sound demodulation.

**Sound Carrier Frequency Offset Recovery:** Both Mono and Stereo IF Carrier frequencies can be adjusted independently (registers [CAROFFSET1](#) and [CAROFFSET2](#)) within a large range (up to 120 kHz for standard mono FM deviations) while the Automatic Standard Recognition System remains active. The frequency offset estimation is written in registers [DC\\_REMOVAL\\_L](#) and [DC\\_REMOVAL\\_R](#) (Mono Left / Channel 1 and Mono Right / Channel 2, respectively) and can be used to implement the Automatic Frequency Control (AFC) via an external I<sup>2</sup>C control.

**Manual Mode:** If required, the Automatic Standard Recognition System system can be disabled (Manual mode) and the user can control all registers including those only controlled by the Automatic Standard Recognition System function when active. Manual mode is selected in register [AUTOSTD\\_STANDARD\\_DETECT](#) (bit LDK\_SCK, I\_SCK, BG\_SCK and MN\_SCK set to 0).



## 4 Dedicated Digital Signal Processor (DSP)

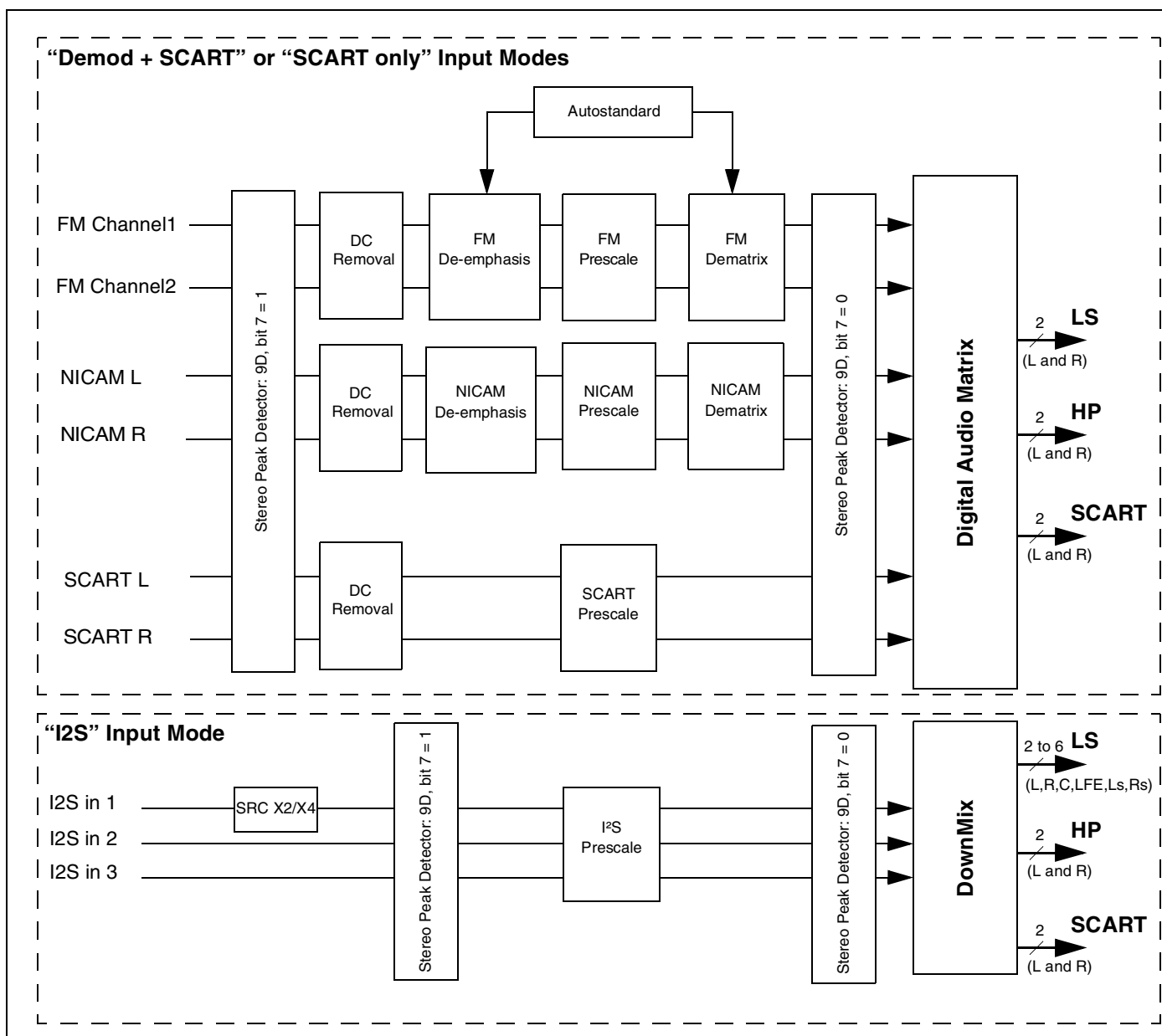
A dedicated Digital Signal Processor (DSP) takes charge of all audio processing features and the low frequency signal processing features of the demodulator. The internal 24-bit architecture will ensure a high quality signal treatment and an excellent dynamic.

### 4.1 Back-end Processing

The “back-end” processing corresponds to the low frequency signal processing (32 kHz or higher frequencies) of the demodulator and other inputs (I<sup>2</sup>S, ADC).

Figure 11 shows a flowchart of the back-end processing tasks. However, the figure shows that the processing is only a SINGLE SOURCE PROCESSING flow (no processing is possible with “Demod + SCART” and I<sup>2</sup>S inputs simultaneously) and that the selection of a headphone output restricts the loudspeakers configuration to 2+1 instead of 5+1.

Figure 11: Back-end Audio Processing



The main features depend on the path:

- FM Channel
  - DC Removal
  - Prescaling
  - De-emphasis (50 or 75 us)
  - Stereo Dematrix
- NICAM Channel
  - DC Removal
  - Prescaling
  - De-emphasis (J17)
  - Dematrix
- Input SCART Channel
  - DC Removal
  - Prescaling
- Input I<sup>2</sup>S Channel
  - I<sup>2</sup>S Prescaling
- Digital Audio Matrix
  - Audio Channel Multiplexer between the different sources (IF, I<sup>2</sup>S, SCART) towards all outputs (S/PDIF, LS, HP or SCART).
- Autostandard management
  - device configuration depending on the standard to be detected
  - freeze the device when a standard is detected
  - once a standard detected, check that there is no change in the detection status
  - set the correct action depending on any change in the detection status (mono backup or mute setup and new standard detection)
- SCART
  - Downmixing:  $L_T / R_T$  or  $L_0 / R_0$  (see AC-3 specification)
  - Soft Mute

## 4.2 Audio Processing

The following software is provided for main loudspeakers (L, R, C, L<sub>S</sub>, R<sub>S</sub>, SubW):

- Downmix
- Dolby® Pro Logic II® Decoder ( $L_T, R_T \rightarrow L, R, C, L_S, R_S, SubW$ ) with Bass Management
- ST WideSurround, ST OmniSurround, SRS® WOW™ or SRS® TruSurround XT® (certified Virtual Dolby® Surround and Virtual Dolby® Digital)
- ST Dynamic Bass
- Smart Volume Control (SVC)
- 5-band Equalizer or Bass-Treble
- Loudness
- Volume with independent channels (Smooth Volume Control)
- Master Volume Control
- Mute/soft-mute

- Balance
- Beeper
- Pink Noise Generator (used to position the loudspeakers)
- Programmable Delay for each loudspeaker
- Adjustable Delay for “lip sync” to compensate audio/video latency up to 60 ms in SCART Only Mode (processing at 48 KHz) and up to 90 ms in Demodulator and SCART Mode (processing at 32 KHz)

**The following software is provided for the headphone or auxiliary output:**

- Downmix
- SRS® TruBass™
- Smart Volume Control (SVC)
- Bass/Treble
- Loudness
- Independent Volume for each channel (Smooth Volume Control)
- Soft Mute
- Balance
- Beeper
- Adjustable Delay for “lip sync” up to 120 ms (to compensate audio/video latency) in SCART Only Mode and up to 180 ms in Demodulator and SCART Mode

**The following software is provided for SCART or S/PDIF outputs:**

- Downmix
- Soft Mute

Figure 12: STV82x7 Audio Processing Flowchart (Front End)

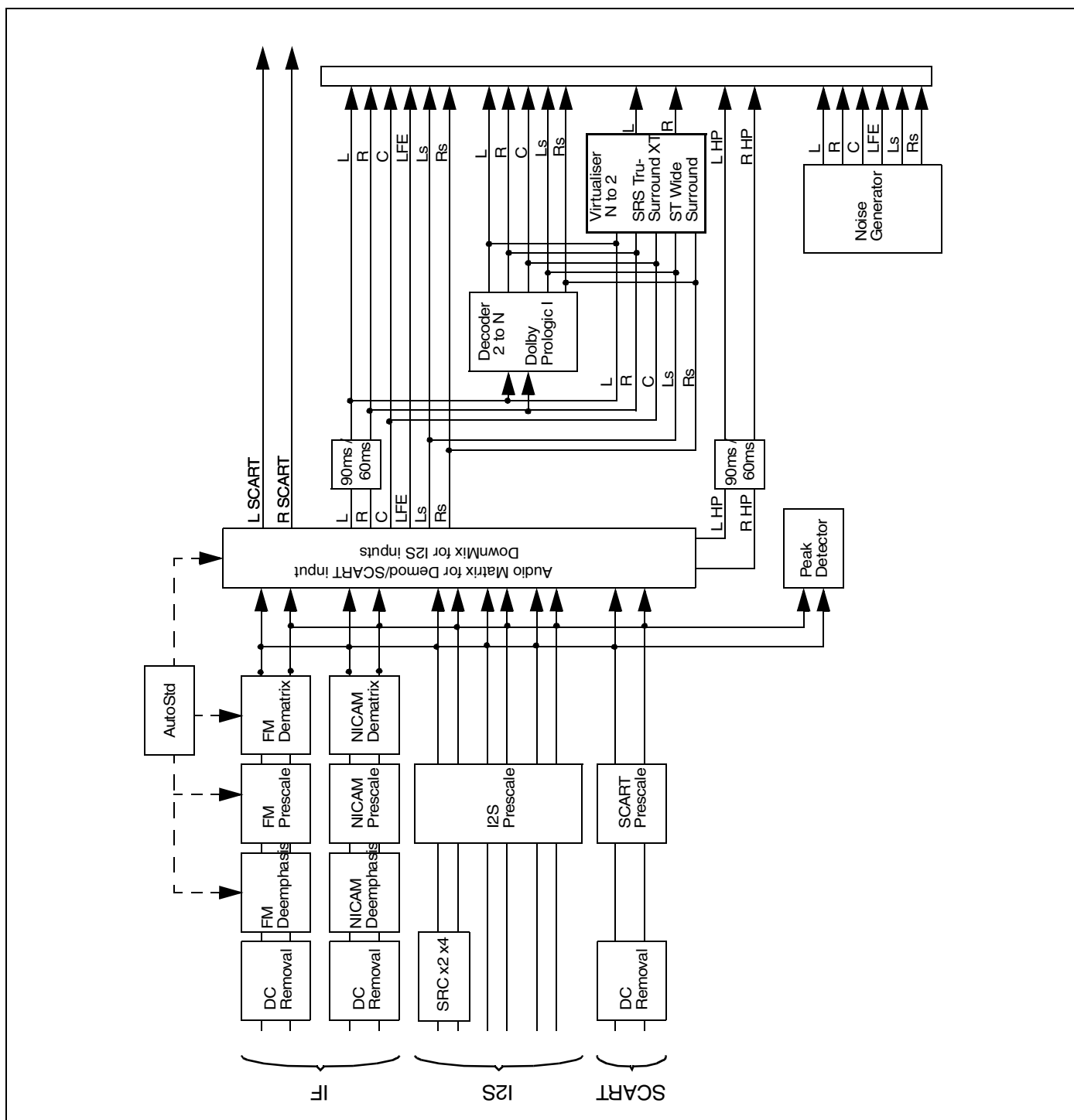
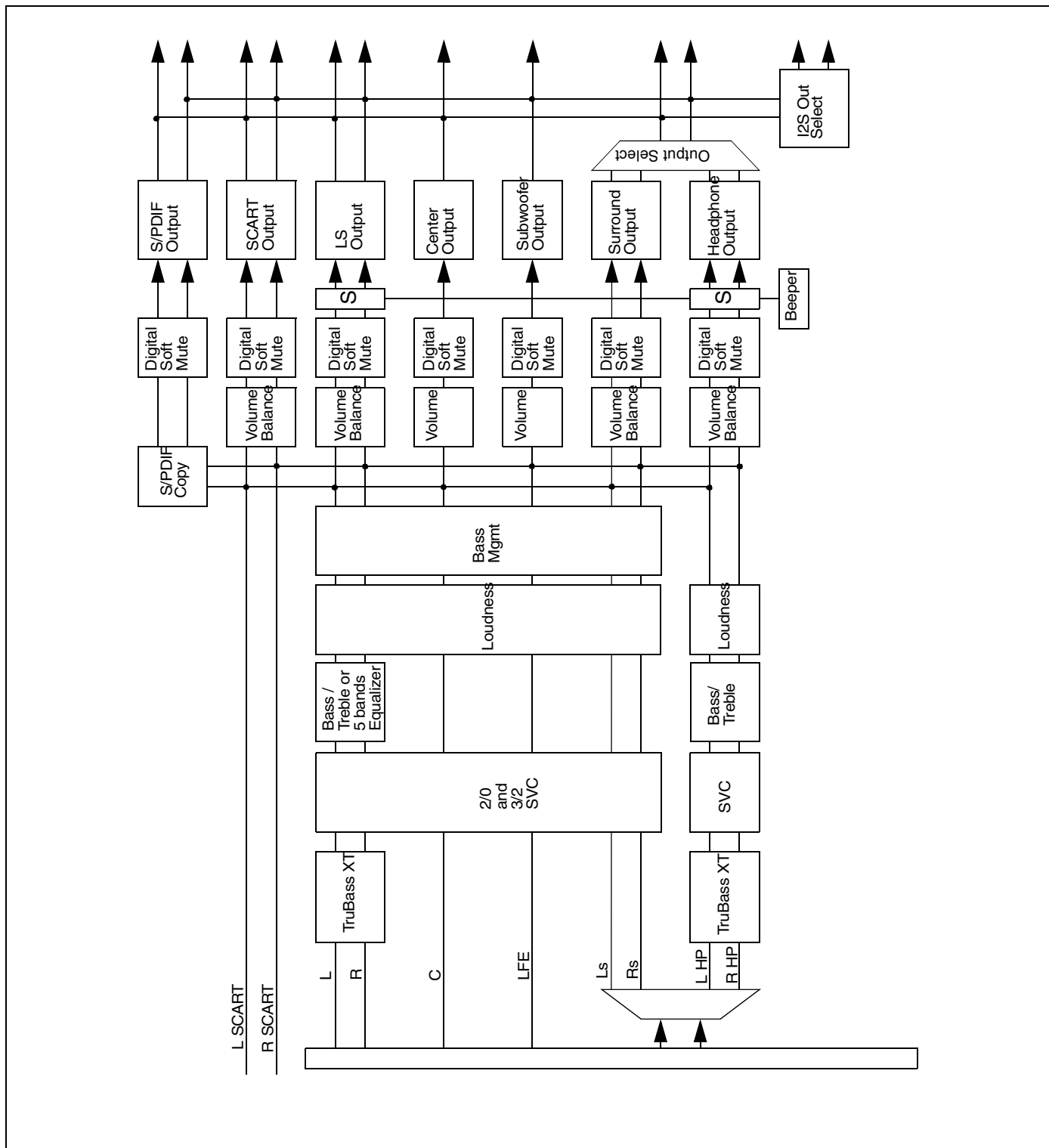


Figure 13: STV82x7 Audio Processing Flowchart (Back End)



### 4.3 ST WideSurround

STV82x7 offers three preset ST WideSurround Sound effects on the Loudspeakers path:

- Music, a concert hall effect
- Movie, for films on TV
- Simulated Stereo, which generates a pseudo-stereo effect from mono source

“ST WideSurround Sound” is an extension of the conventional stereo concept which improves the spatial characteristics of the sound. This could be done simply by adding more speakers and coding more channels into the source signal as is done in the cinema, but this approach is too costly for normal home use. The ST WideSurround system exploits a method of phase shifting to achieve a similar result using only two speakers. It restores spatiality by adding artificial phase differences.

The Surround/Pseudo-stereo mode is automatically selected by the Automatic Standard Recognition System (Autostandard) depending on the detected stereo or mono source. By default, “Movie” is selected for Surround mode. This value may be changed to “Music” by the STSRND\_MODE bit in the [STSRND\\_CONTROL](#) register.

Additional user controls are provided to better adapt the spatial effect to the source. The ST WideSurround Gain ([STSRND\\_LEVEL](#)) and ST WideSurround Frequency ([STSRND\\_FREQ](#)) registers can be used to enhance Music Predominancy in Music mode and Theater effect and Voice Predominancy in Movie mode.

### 4.4 ST OmniSurround

STV82x7 offers a spatial virtualizer to output any multi-channel input in stereo on the Loudspeakers path:

“ST OmniSurround” will recreate a multi-channel spatial sound environment using only the Left and Right front speakers. It can be adapted to any input configuration (OMNISRND\_INPUT\_MODE).

ST Voice will allow you to enhance the voice content of your program to increase the intelligibility and the presence of the sound.

### 4.5 Dolby Pro Logic II Decoder

Dolby® Pro Logic II® is a matrix decoder that decodes the five channels of surround sound that have been encoded onto the stereo sound tracks of Dolby® Surround program material such as DVD movies and TV shows.

It is even possible to decode standard stereo signals like music or non encoded movies. Furthermore, it is an active process designed to enhance sound localization through the use of very high-separation decoding techniques.

The Dolby® Pro Logic II® decoder is also able to emulate the former Dolby® Pro Logic® decoder in a specific mode.

### 4.6 Bass Management

This processing will generate the subwoofer signal and adjust all loudspeakers channels gain and bandwidth.

Speakers capable of reproducing the entire frequency range will be referred to as “full range speakers”, then signals sent to full range speaker will be full bandwidth (no filtering).

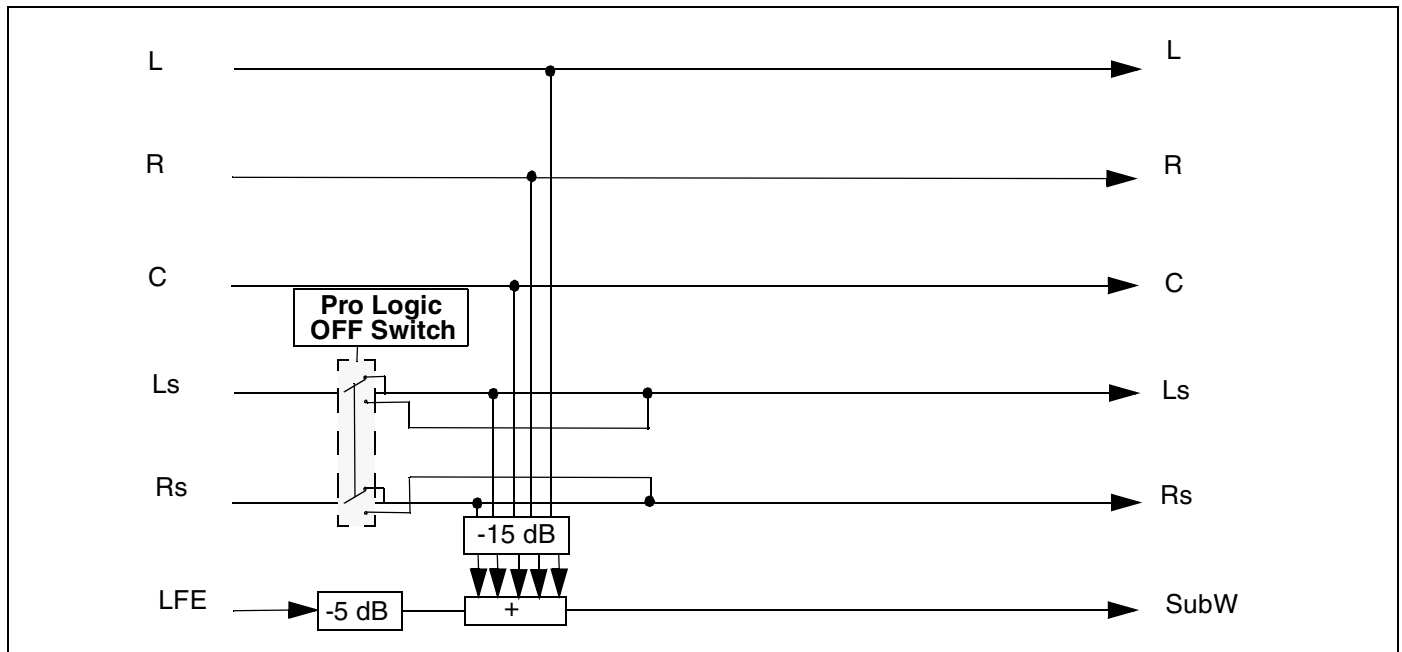
Speakers that have limited bass handling capabilities will be referred to as “satellite speakers”, then signals sent to satellite speaker will be high-pass filtered to remove bass information below 100 Hz.

In the STV82x7, five output configuration modes have been implemented according to “Dolby Digital Consumer Decoder” specifications. They are described below.

#### 4.6.1 Bass Management Configuration 0

In some cases, the bass management filters are available in the decoder itself, so there is no need to reproduce these filters. The output configuration shown in [Figure 14](#) offers this possibility.

**Figure 14: Bass Management Configuration 0 (with Pro Logic switch indicating its reset state)**

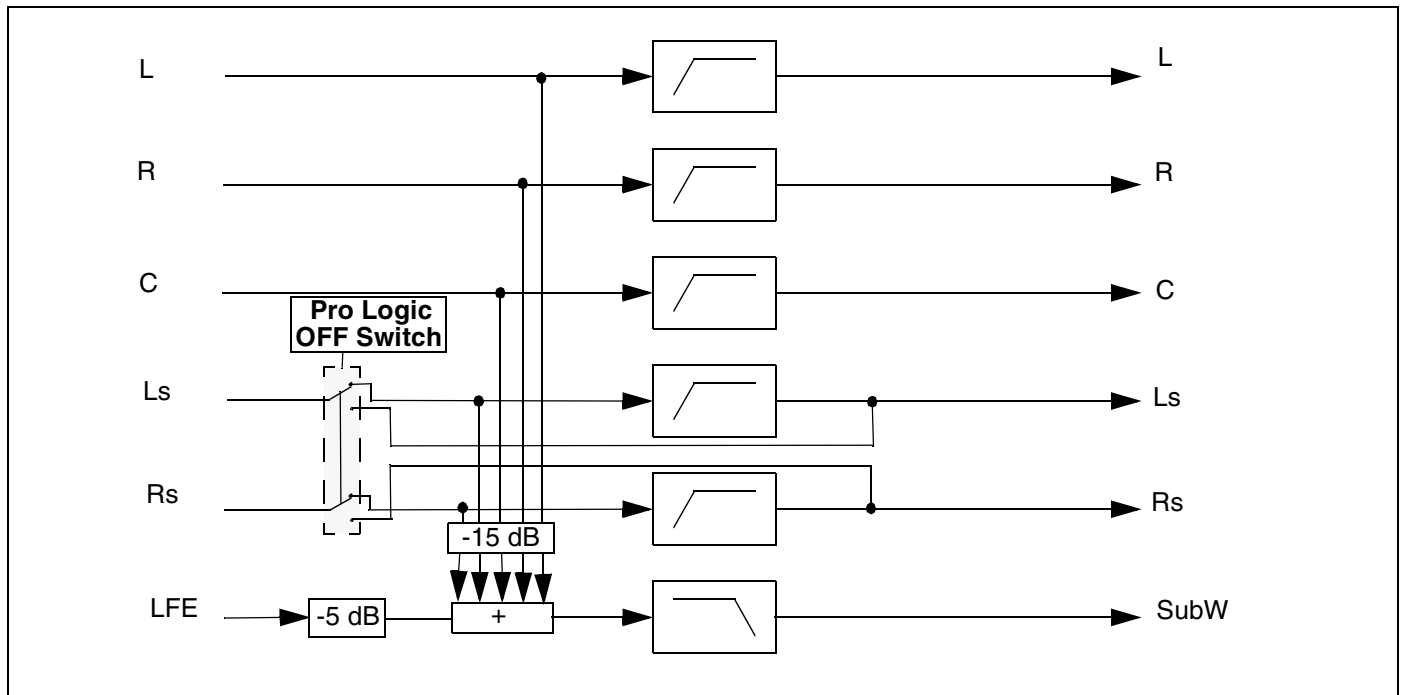


#### 4.6.2 Bass Management Configuration 1

Configuration 1, shown in Figure 15, assumes that all five speakers are not full range and that all of the bass information will be redirected to and reproduced by a single subwoofer. This configuration is intended for use with 5 satellite speakers.

To prevent signal overload, the five main channels are attenuated by 15 dB, while the LFE channel is attenuated by 5 dB to maintain the proper mixing ratio.

**Figure 15: Bass Management Configuration 1 (with Pro Logic switch indicating its reset state)**



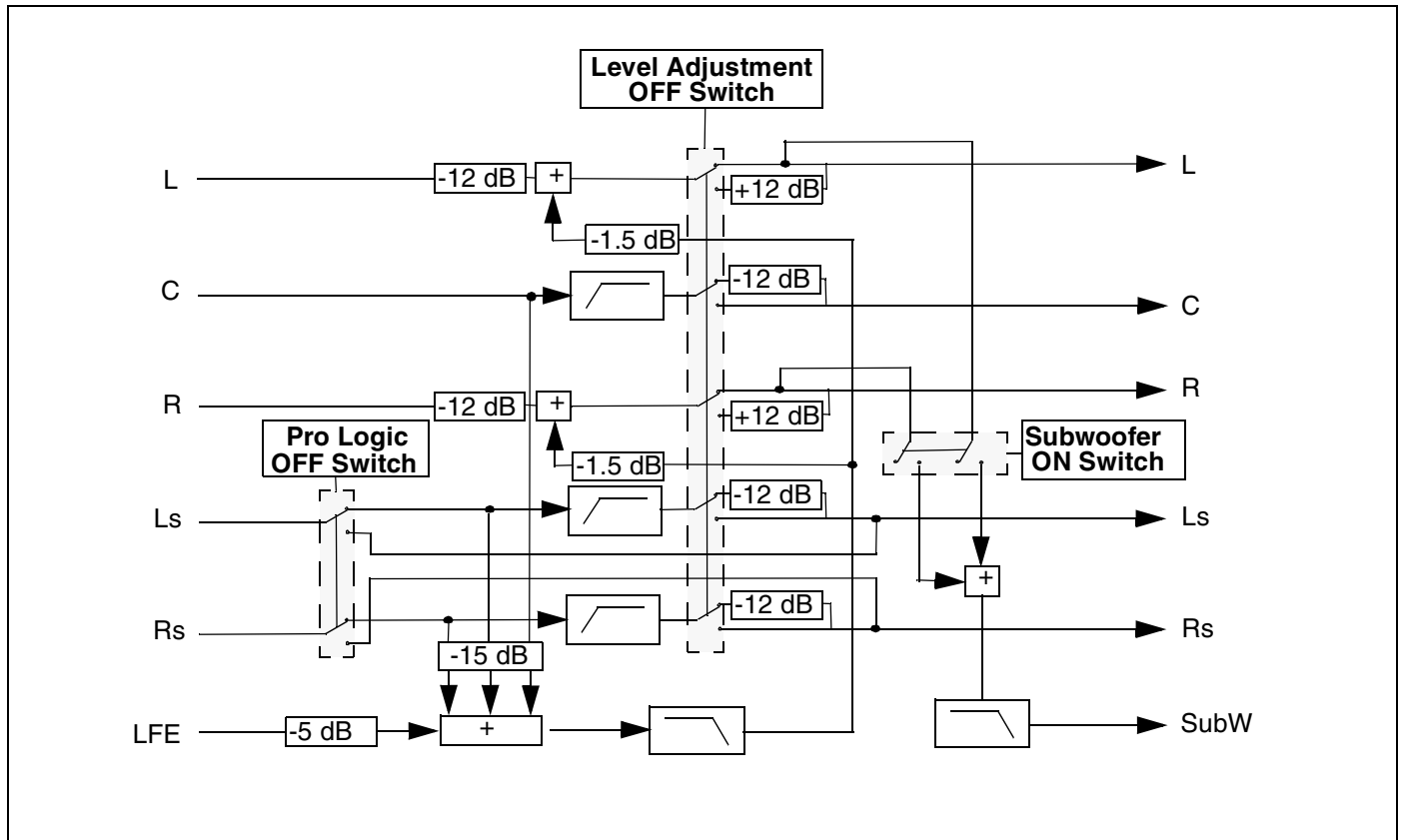


### 4.6.3 Bass Management Configuration 2

Configuration 2 assumes that the left and right speakers, are full range while the center and surround speakers are smaller speakers. Also, all bass data is redirected to the left and right speakers.

This configuration include output level adjustment that allows 12 dB attenuation for the 3 smaller speakers (C, Ls, Rs). When the level adjustment will be disabled the decoder boosts by 12 dB the full range speakers (Left, Right).

Figure 16: Bass Management Configuration 2 (all switches indicate their reset state)

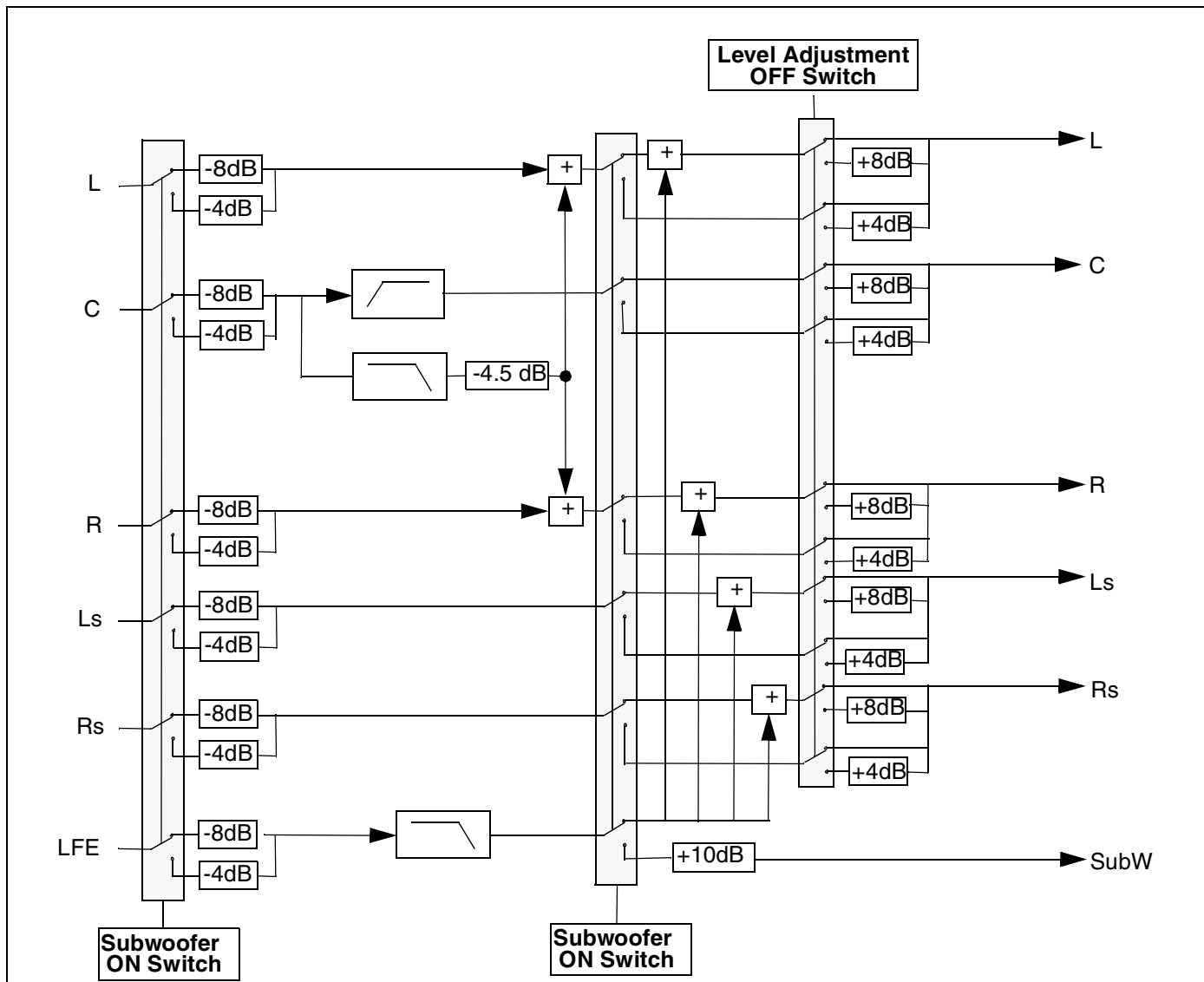


### 4.6.4 Bass Management Configuration 3

The third configuration, shown in Figure 17, assumes that all speakers except the center are full range, then all bass information will be directed to and reproduced by the front left and front right and both surround speakers. In order to provide more flexibility to this configuration, a switch will offer an option which will produce a subwoofer channel by the LFE channel.

When the Subwoofer Switch is OFF, the input channels will be attenuated by 8 dB. Configuration 3 is required in certain high-end products.

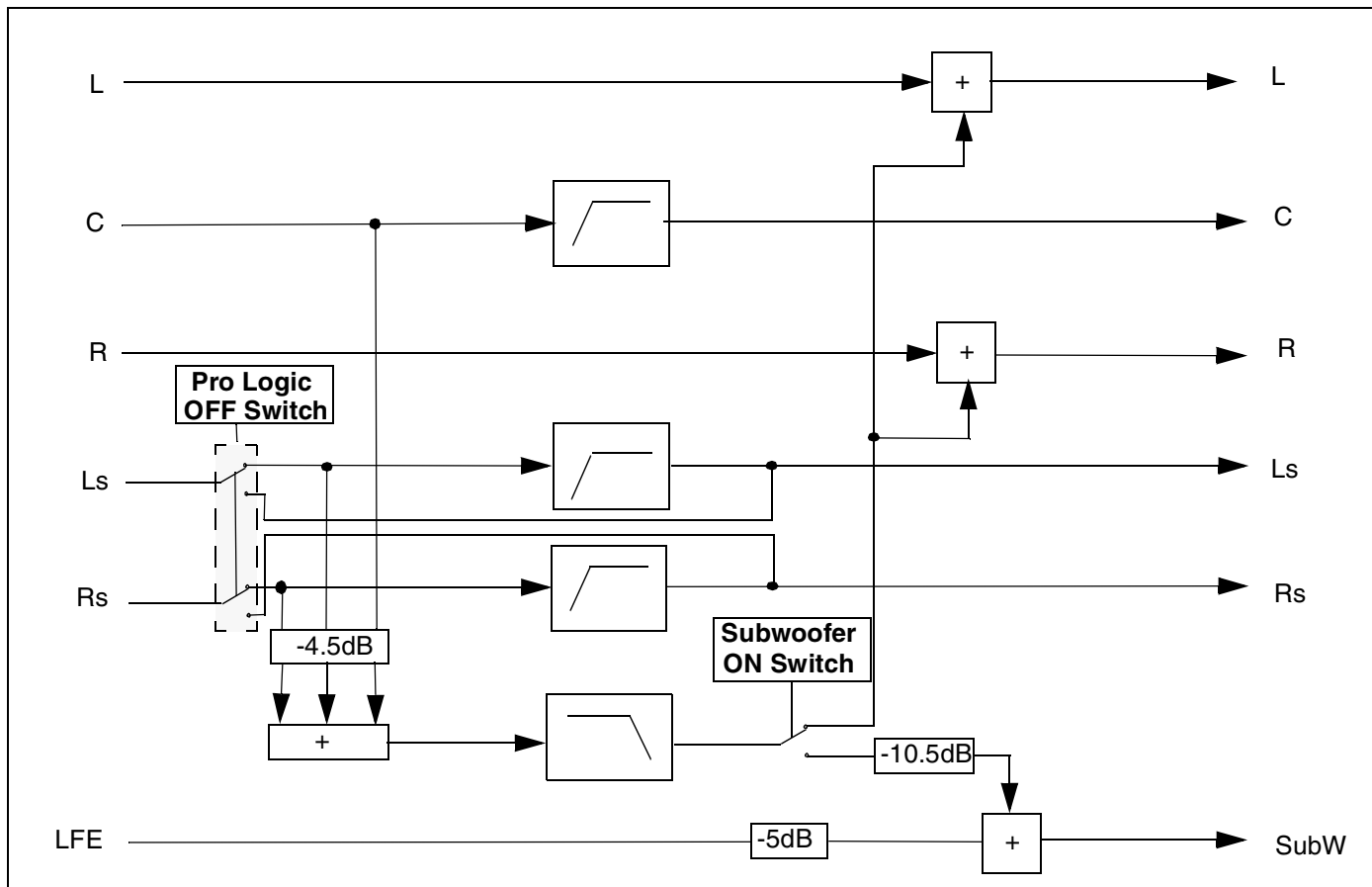
Figure 17: Bass Management Configuration 3 (all switches indicate their reset state)



### 4.6.5 Bass Management Configuration 4

This configuration implements the Simplified Dolby configuration. The center, left surround and right surround channels are summed and then filtered by the LPF. The composite bass information is either summed back into the left and right channels or summed with the LFE channel and sent to the subwoofer output, see Figure 18.

Figure 18: Implementation of the Bass Management Configuration 4 (Simplified Configuration)



### 4.7 SRS WOW and TruSurround XT

The SRS® TruSurround XT™ is a processing system that can accept from 1 to 6 channels on input and that will generate a 2-channel output signal.

This processing system includes the latest SRS® algorithms:

- SRS® WOW™
- SRS® TruSurround® (Multi-channel signal virtualizer)

#### 4.7.1 SRS TruSurround

The SRS® TruSurround® is a processing that can accept from 2 to 5 channels on input and that will generate a 2-channel output signal.

SRS® TruSurround® uses Head-Related Transfer Function (HRTF) -based frequency tailoring of (L/R) difference signals to extend the sound image out past the physical boundaries of the speaker placements to surround channel information. These rear channel HRTF curves have much greater peak to valley differences at center frequencies. These were chosen to cause rear channel difference signals to virtualize farther behind the listener and directed to a different virtual position as compared to front channel signals. Information that is equal (L+R) in the rear surround channels

is processed by an identical HRTF curve but mixed in at a much lower amount. This HRTF processing of equal (L/R) signals was again used to virtualize information to the rear of the listener.

The SRS® TruSurround® is certified by Dolby Laboratories to be a Virtual Dolby® Digital and Virtual Dolby® Surround.

#### 4.7.2 SRS WOW

The SRS® WOW™ is an a sound processing system including:

- SRS® 3D Mono/Stereo™
- SRS® Dialog Clarity™
- SRS® TruBass™

##### 4.7.2.1 SRS 3D Mono/Stereo

This system is used to create a pseudo-stereo signal for mono inputs or a three-dimensional spatial signal for stereo inputs.

##### 4.7.2.2 SRS Dialog Clarity

This system is used to enhance dialog perception.

##### 4.7.2.3 SRS TruBass

The SRS® TruBass™ audio enhancement technology provides deep, rich bass to small speaker systems without the need for a subwoofer or additional extra physical components. For systems with a subwoofer, TruBass™ complements and enhances bass performance. Psycho-acoustically, when the human ear is presented with a low frequency sound signal that is missing the fundamental harmonic, it will fill in the fundamental frequency based on the higher harmonics that are present. By accentuating the second and higher frequency harmonics of the bass portion of a signal, TruBass™ gives the perception of greatly improved bass response.

SRS® TruBass™ is implemented on loudspeakers path, headphone path or on both in parallel.

## 4.8 Smart Volume Control (SVC)

The Smart Volume Control regulates the audio signal level before audio processing. This regulation is necessary in order for the signal level to be independent from the source (terrestrial channels, I2S or SCART), its modulation (AM, FM or NICAM) and annoying volume changes (advertising, etc.). The Smart Volume Control works as an audio compressor/expander; i.e. when the input signal exceeds the threshold level, a very rapid attenuation (-2 dB/ms) is applied to rescale the signal down to the threshold value. When the input signal is below the threshold level, the previous attenuation is reduced slowly in order to retrieve the original input level (0 dB gain). If the input signal is too low, an addition gain of 6 dB can be provided.

To personalize the action of the SVC, five parameters are available:

1. Threshold: Maximum quasi-peak level that can be expected on output
2. Peak measurement mode: Select the channel on which the peak measurement must be performed (Left, Right, Center...)
3. Release time: Gain slope applied to the amplification phase
4. Expander switch: To allow a +6dB amplification of small signals in order to reduce the output dynamic range
5. Make up gain: Allows compensation of the signal amplitude limitation thanks to a 0 to 24 dB adjustable gain.

The SVC is implemented on the loudspeakers path, headphone path or on both in parallel (independent settings). Also, the SVC can be applied in six-channel mode (L, R, L<sub>S</sub>, R<sub>S</sub>, C and SubW).

## 4.9 ST Dynamic Bass

STV82x7 offers dynamic bass boost processing on the Loudspeakers path:

ST Dynamic Bass is a bass boost process that can dramatically increase the bass content of any program without any output level saturation.

3 cutoff frequencies (BASS\_FREQ) can be chosen, 100 Hz, 150 Hz and 200 Hz to adapt the effect to your loudspeakers. The amount of bass (BASS\_LEVEL) can also be fine tuned in order to adapt the effect loudness.

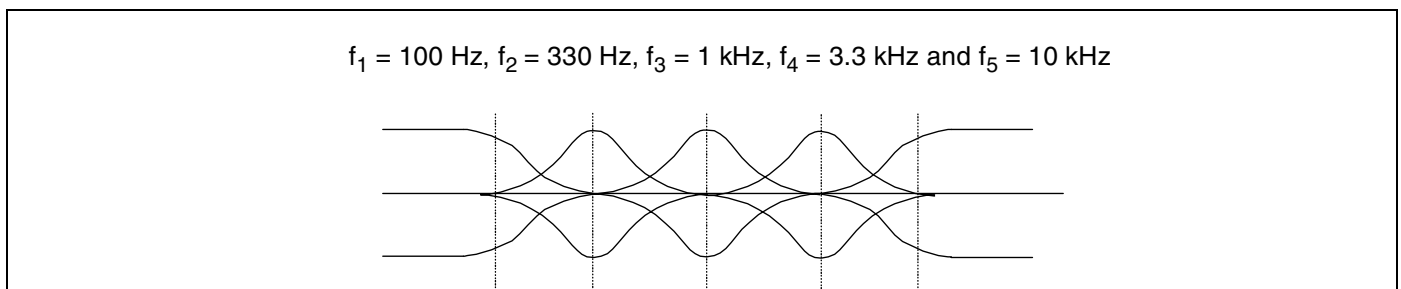
## 4.10 5-Band Audio Equalizer

The loudspeakers audio spectrum is split into 5 frequency bands and the gain of each of band can be adjusted within a range from -12 dB to +12 dB in steps of 0.25 dB. The Audio Equalizer may be used to pre-define frequency band enhancement features dedicated to various kinds of music or to attenuate frequency resonances of loudspeakers or the listening environment. The Equalizer is enabled by the LS\_EQ\_ON bit in the [LS\\_EQ\\_BT\\_CTRL](#) register. The gain value for Band X is programmed in register [EQ\\_BANDX\\_GAIN](#).

The 5-Band Audio Equalizer is exclusive with Bass-Treble control. Bit LS\_EQ\_BT\_SW in register [LS\\_EQ\\_BT\\_CTRL](#) is used to select either the 5-Band Audio Equalizer or the Bass-Treble control for the Loudspeakers path.

Depending on the LS Equalizer or LS Bass-Treble value, the volume level can be clamped to the LS output to prevent any possible signal clipping from occurring using the ANTICLIP\_LS\_VOL\_CLAMP bit in the [VOLUME\\_MODES](#) (D7h) register.

Figure 19: Equalizer



## 4.11 Bass/Treble Control

The gain of bass and treble frequency bands for Headphone can be also tuned within a range from -12 dB to +12 dB in steps of 0.25 dB. It may be used to pre-define frequency band enhancement features dedicated to various kinds of music. The Headphone Bass/Treble feature is enabled by setting the HP\_BT\_ON bit in the [HP\\_BT\\_CONTROL](#) register. The Bass and Treble gain values are adjusted in registers [HP\\_BASS\\_GAIN](#) and [HP\\_TREBLE\\_GAIN](#), respectively.

Depending on the HP Bass-Treble value, the volume level can be clamped to the HP output to prevent any possible signal clipping from occurring using the ANTICLIP\_HP\_VOL\_CLAMP bit in the [VOLUME\\_MODES](#) (D7h) register.

## 4.12 Automatic Loudness Control

As the human ear does not hear the audio frequency range the same way depending on the power of the audio source, the Loudness Control corrects this effect by sensing the volume level and then boosting bass and treble frequencies proportionally to middle frequencies at lower volume.

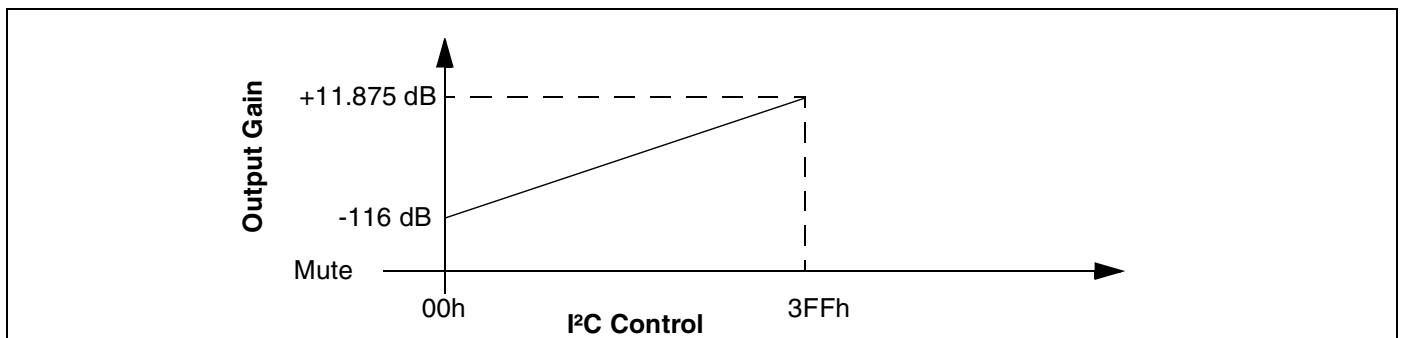
While maintaining the amplitude of the 1 kHz components at an approximately constant value, the gain values of lower and higher frequencies are automatically progressively amplified up to +18 dB when the audio volume level decreases. The maximum treble amplification can be adjusted from 0 dB (first order loudness) to +18 dB (second order loudness) in steps of 3 dB. As the volume is proportional to the external audio amplification power, the loudness amplification threshold is programmable in order to tune the absolute level. The Loudspeakers Loudness function is enabled by setting the LS\_LOUD\_ON bit in register [LS\\_LOUDNESS](#). The Loudspeakers Loudness Threshold and Maximum Treble Gain values are also programmed in this register. The Headphone Loudness function is enabled by setting the HP\_LOUD\_ON bit in register [HP\\_LOUDNESS](#). The Headphone Loudness Threshold and Maximum Treble Gain values are also programmed in this register.

The loudness cut-off frequency is 100 Hz.

## 4.13 Volume/Balance Control

The STV82x7 provides a Volume/Balance Control for all output channels configuration (except for S/PDIF) with different volume level per channel (L, R, C, L<sub>S</sub>, R<sub>S</sub>, SubW, SCART). Its wide range (from +11.875 to -116 dB, in a dB linear scale with a 0.125 dB step) largely covers typical home applications (approx. 60 dB) while maintaining a good S/N ratio.

Figure 20: Volume Control



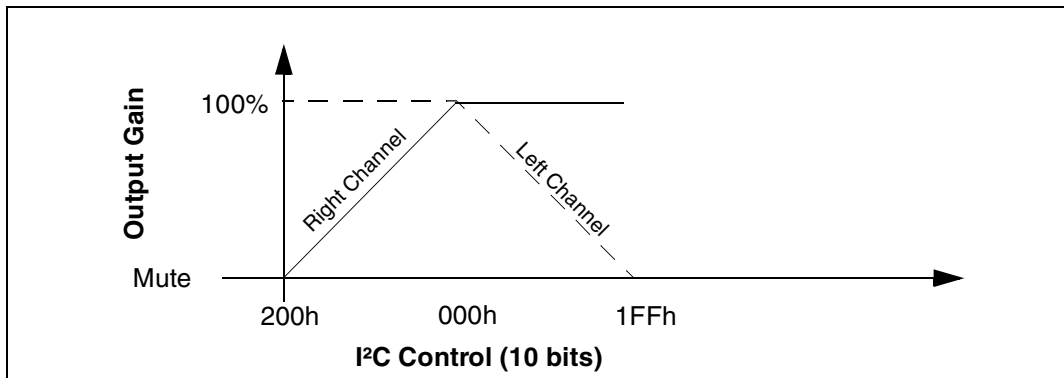
An extra Master Volume Control can apply an extra gain/attenuation on L, R, C, L<sub>S</sub>, R<sub>S</sub> and SubW channels.

The Volume/Balance Control can operate in one of two different modes:

- In **Differential mode** (default value), the volume control is a common volume value for both the Left and Right Loudspeakers or Headphone channels (see [Figure 20](#)) and complimentary balance control is used (see [Figure 21](#)).

- In **Independent mode**, the volume for the Left and Right channels for Loudspeakers or Headphone is controlled independently.

**Figure 21: Differential Balance**



Note: Each step is 0.25 dB

#### 4.14 Soft Mute Control

The Digital Soft Mute is applied smoothly (20 ms for 120 dB range) to avoid any switch noise on output. It is available on all output channels pairs:

- S/PDIF channel (Left/Right)
- SCART channels (Left/Right)
- Loudspeakers channels (Left/Right)
- Center
- Subwoofer
- Headphone/Surround channels (Left/Right)

Another soft mute (analog) is also available on each DAC output.

#### 4.15 Beeper

The beeper is used to generate a tone on the Loudspeakers or/and Headphone outputs. The beeper sound (square wave) is added to the audio signal which is attenuated by 20 dB. The beep sound amplitude includes a smooth attack and decay to avoid any parasitic noise when starting and stopping.

It can be used for various applications such as beep sounds for remote control, alarm clock or other features.

The Beeper operates in one of two modes:

- **Pulse mode** (beep applications): A tone with a programmable short duration (0.1, 0.25, 0.5 and 1.0 s) is generated. Afterwards, the beeper is automatically disabled and the output is switched back to the audio signal, see [Figure 22](#).
- **Continuous mode** (alarm application): A tone with a programmable long duration is generated. Its start and stop controls must be programmed by I²C, see [Figure 23](#).

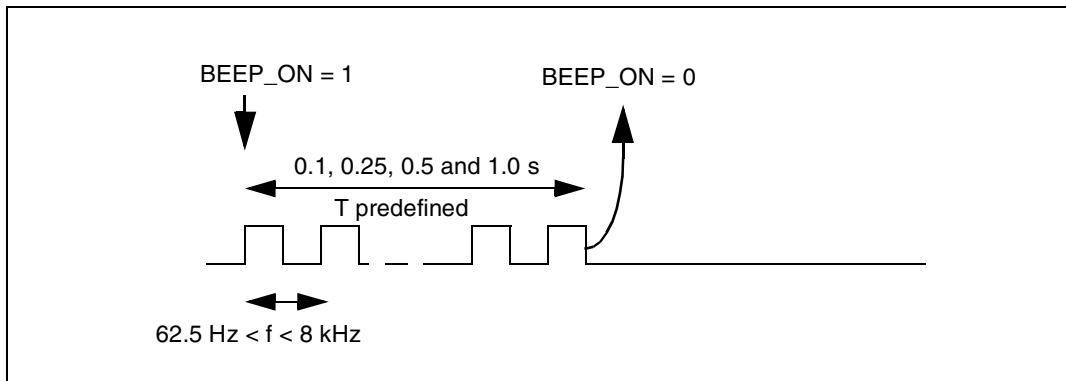
The Beeper function is enabled by setting the BEEPER\_ON bit in register [BEEPER\\_ON](#).

Beeper parameters are controlled in register [BEEPER\\_MODE](#).

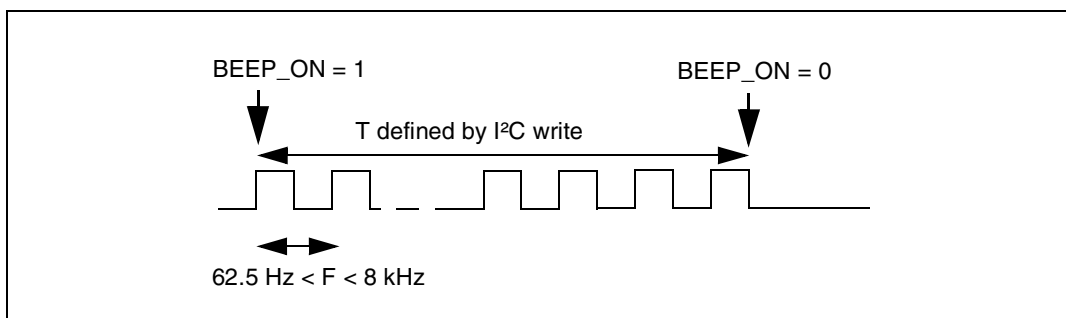
The beeper tone level and frequency are programmed in register `BEEPER_FREQ_VOL`. The level (or volume) ranges between 0 dB and -93 dB in steps of 3 dB and the tone frequency ranges between 62.5 Hz and 8 kHz in steps of 1 octave.

A beep generator is shared only by the Loudspeakers or Headphone outputs. Therefore, in the event of simultaneous beeps when in Pulse mode, only the first beep will define the effective duration that will be the same for both outputs.

**Figure 22: Pulse Mode**



**Figure 23: Continuous Mode**



## 4.16 Internal Audio/Video Delay (Lip Sync)

Since increasing processing on the video signal implies more delay compared to the audio signal, there is a possibility inside the device of compensating by inserting a delay on the audio path in order to resynchronize the two signals:

- 60ms with 48 KHz sampling frequency (SCART only input mode)
- 90ms with 32 KHz sampling frequency (demodulator input mode)

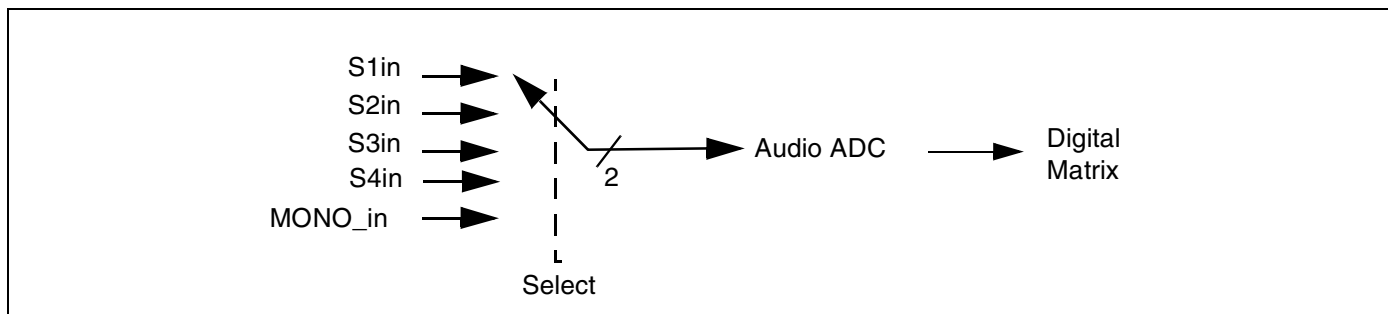
The same delay is available for the LS path and/or the HP path.



## 5 Analog Audio Matrix (In / Out)

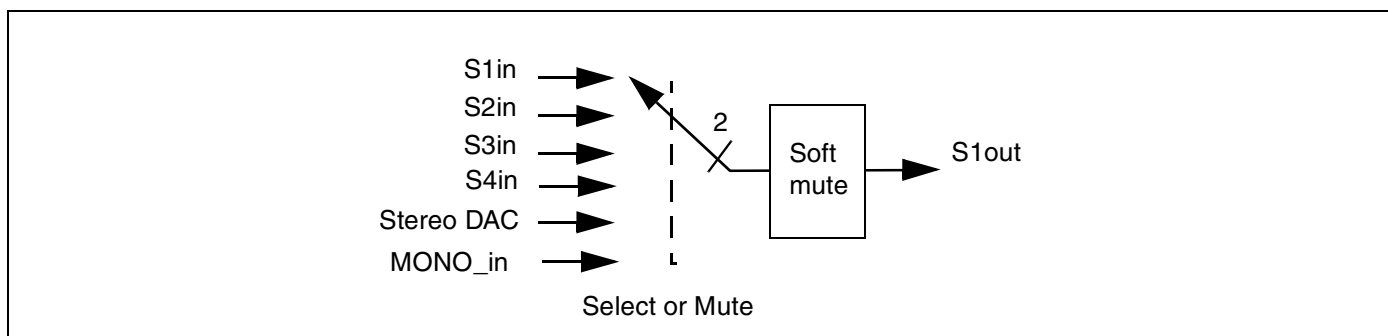
The analog part of the audio matrix can be divided into two parts: the SCART input matrix and the SCART output matrix.

Figure 24: SCART Input Matrix



The SCART input matrix is an input for the digital matrix (after the ADC) which select which source will be sent to the DSP.

Figure 25: SCART1/2/3 Output Matrix



The SCART output matrix selects the sound to output, which can be directly a SCART input or the output of the DSP. A mute function is provided to switch off the outputs.

A soft-mute function is provided to avoid all spurious sounds when switching from one position to another position.

The SCART 2 and 3 output matrices have the same functions as the SCART 1 output matrix.

The particularity of the matrix is to accept input signal of  $2 V_{RMS}$  and to have the capability to output such level. In this case, the power supply must be 8 V.

**The Mono audio input is able to accept signals with a  $0.5 V_{RMS}$  amplitude.**

## 6 I<sup>2</sup>S Interface (In / Out)

The STV82x7 offers three input/output choices: one I<sup>2</sup>S input, three I<sup>2</sup>S inputs or one I<sup>2</sup>S output.

### 6.1 I<sup>2</sup>S Inputs

The STV82x7 can interface with a digital sound decoder. In this case, the digital data can be input at a speed of 0.384 Mbytes/s (3.072 MHz for a 48 kHz sampling frequency with 32 bits of data). In compliance with Dolby® specifications, only the sampling frequency is subject to restrictions. All other requirements are extracted from other various specifications.

**Table 5: I<sup>2</sup>S Characteristics**

<b>Sampling Frequency (kHz)</b>	8, 11.025, 12,16, 22.05, 24, 32, 44.1 and 48
<b>Data Size</b>	16, 18 <sup>*</sup> , 20 <sup>*</sup> , 24 <sup>*</sup> , 32
<b>PCMCLK</b>	$512 \times f_s^{1\ 2}$

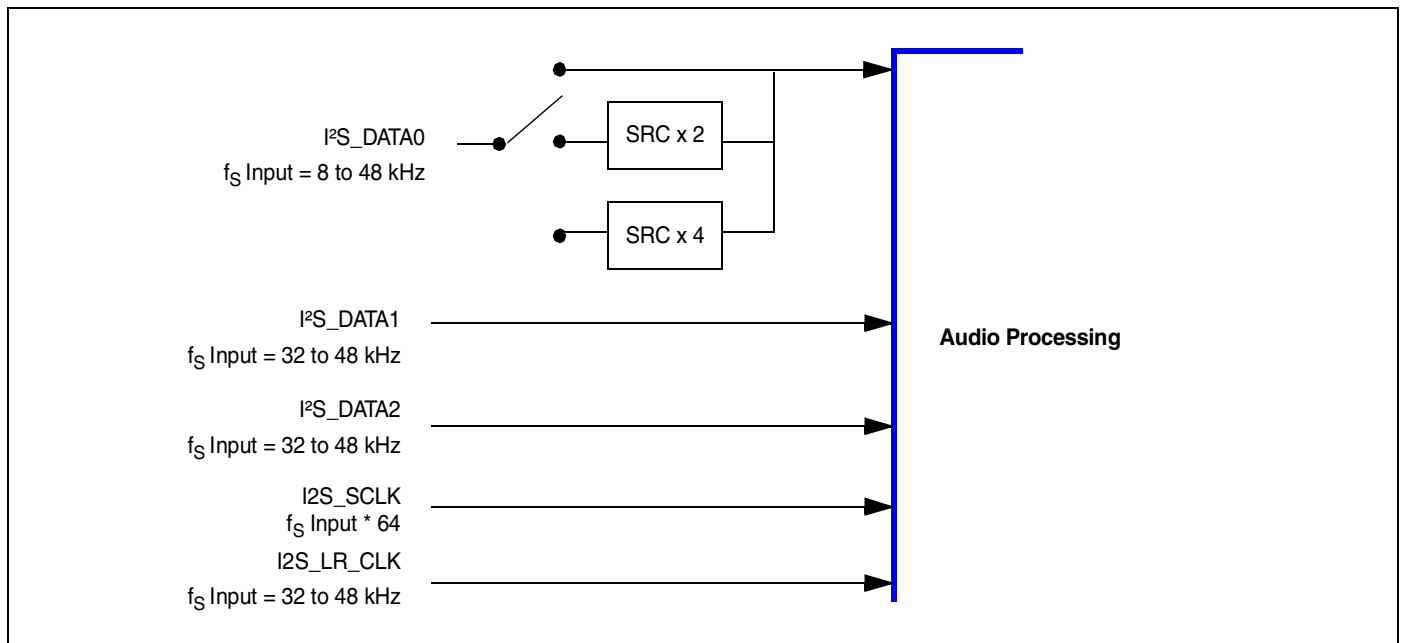
1. means that the number is the number of effective bits but the transmission is with 32 bits.
2.  $512 \times f_s$  is used by the DACs if  $512 \times f_s$  is present.

The PCMCLK (possible clock for upsampling) is provided by the master which is the digital sound decoder. A sample rate conversion (SRC) will be necessary in the second case (STV82x7 slave) in order to have a fixed frequency output from this block (either 32 kHz, 44.1 kHz or 48 kHz).

*Note:* The SRC function is only available in single I<sup>2</sup>S input mode.

The I<sup>2</sup>S interface is used in two ways depending on the package:

1. The interface with one I<sup>2</sup>S (I<sup>2</sup>S\_DATA0) connection (only stereo or stereo-coded Dolby® Pro Logic®);
2. One interface with three I<sup>2</sup>S connections connected to the DSP to allow the processing of a multi-channel signal (maximum of 6 channels).

Figure 26: I<sup>2</sup>S Block Diagram

- Note: 1 The I<sup>2</sup>S input and output modes are exclusive (this means that the I2S\_DATA0 can be used as input or as output).*
- 2 Simultaneous processing of I<sup>2</sup>S inputs and SIF inputs and/or ADC inputs (SCART or MONO inputs) is NOT possible with the device.*
- 3 I2S\_PCM\_CLK is not needed for the device.*

Table 6: I<sup>2</sup>S Frequency Configuration

I <sup>2</sup> S (Max. Number of Channels)	f <sub>S</sub> Input (kHz)	f <sub>S</sub> Output (kHz) after SRC	SRC Use
1 (I <sup>2</sup> S_DATA0)	8	32.0	x 4
1 (I <sup>2</sup> S_DATA0)	16	32.0	x 2
3	32	32.0	No
1 (I <sup>2</sup> S_DATA0)	11.025	44.1	x 4
1 (I <sup>2</sup> S_DATA0)	22.05	44.1	x 2
3	44.1	44.1	No
1 (I <sup>2</sup> S_DATA0)	12	48.0	x 4
1 (I <sup>2</sup> S_DATA0)	24	48.0	x 2
3	48	48.0	No

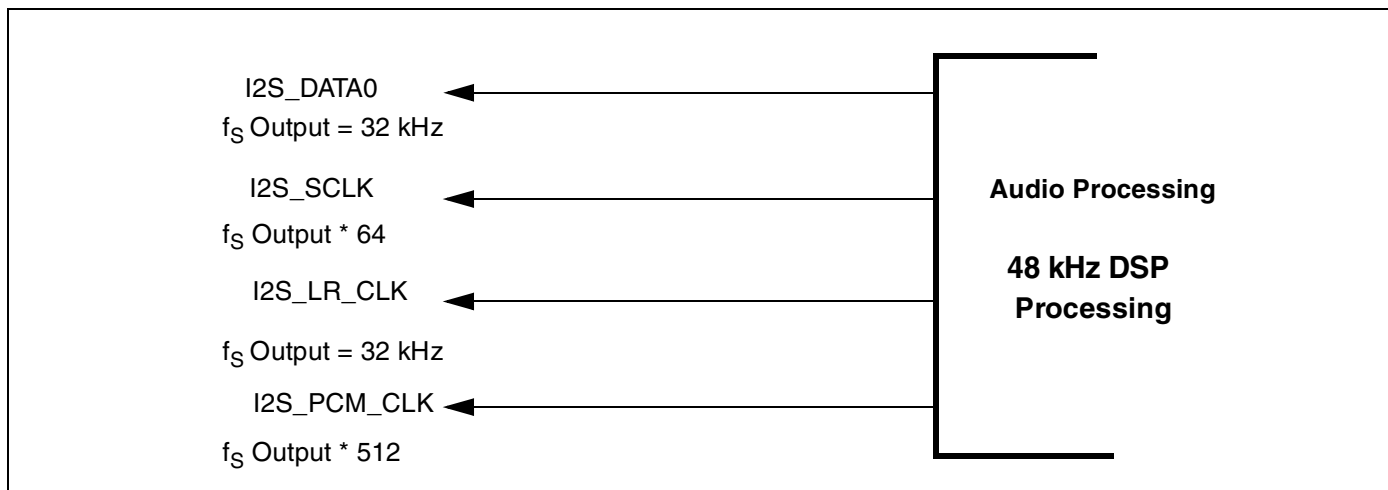
Both standard and non-standard modes are available, see [Figure 29](#).

## 6.2 I<sup>2</sup>S Output

A digital stereo output (I<sup>2</sup>S compatible) is also available for routing the demodulated signal or a converted input audio signal to an external device. In this case, the I2S\_DATA0 signal and all clock signals are set as outputs by setting bit D6 in register RESET to 1.

The STV82x7 I<sup>2</sup>S drives the serial bus (SCLK, LR\_CLK, I<sup>2</sup>S\_DATA0) in master mode in 64.fs format with a sampling frequency ( $f_s$ ) of 32 kHz. The I<sup>2</sup>S\_PCM\_CLK signal can be used as a master clock in 512.fs format if required for the slave interface. Both standard and non-standard modes are available, see [Figure 29](#).

Figure 27: TQFP 80 I<sup>2</sup>S Output Block Diagram



*Note:* The I<sup>2</sup>S input and output modes are exclusive (this means that the I2S\_DATA0 can be used as input or as output).

Figure 28: I<sup>2</sup>S Output Selection

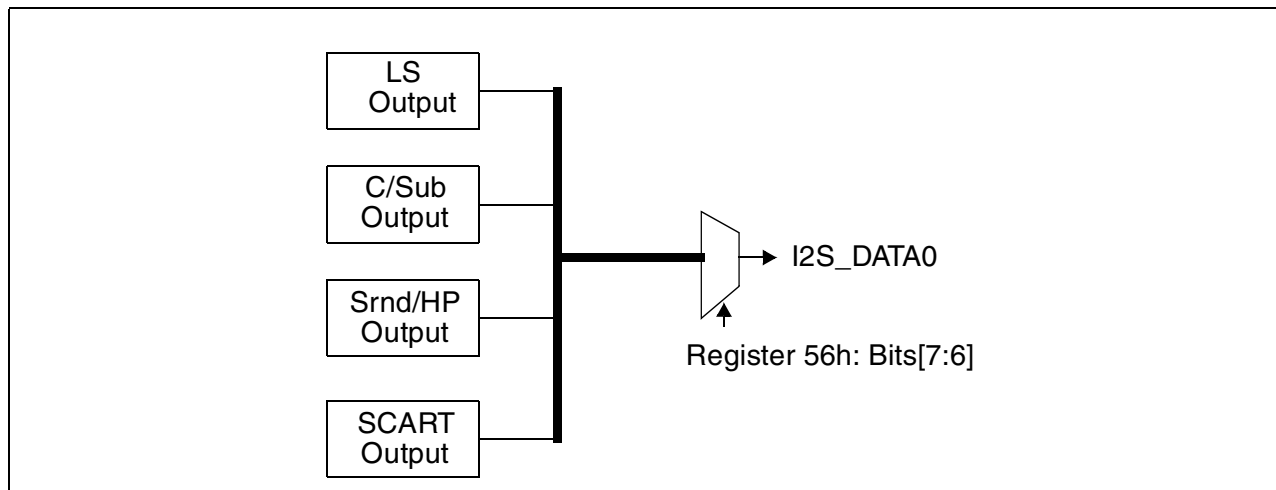
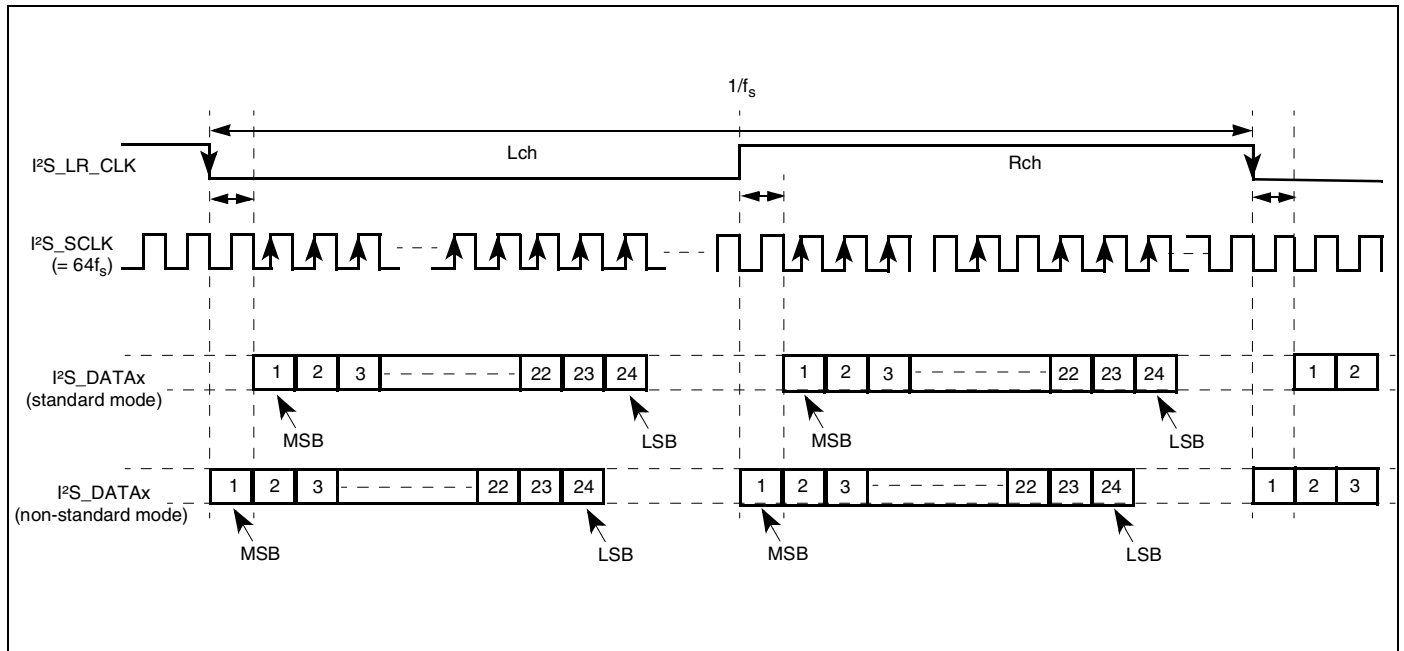


Figure 29: I<sup>2</sup>S Data Format: Lch = LOW, Rch = HIGH (I<sup>2</sup>S Input or Output mode)



## 7 S/PDIF Input/Output

An S/PDIF output is available for connection with an external A/V decoder/amplifier.

The signal on this S/PDIF output is selected by an on chip Multiplexer between the internal signal and an external signal present on S/PDIF bypass input (Pin 44) with SPDIF\_MUX bit in the DAC\_CONTROL register.

The outputted internal signal can be selected from:

- L/R
- C/Subwoofer
- HP or Surround L/R
- SCART L/R

The external signal is for example the signal provided by an external **Dolby® Digital** decoder (STD2000).

Mute facility is also provided on the S/PDIF output.

*Note: The S/PDIF\_IN pin (Pin 44) is a CMOS digital pin and input signals on this pin must fulfill the characteristics as mentioned in [Section 13.12: Digital I/Os Characteristics on page 144](#) ( $\pm 0.5 V_{PP}$  standard S/PDIF input level is not directly supported by the device and needs external circuitry).*

## 8 Power Supply Management

A mixed supply voltage environment requires the following voltages:

- 3.3 V capable inputs/outputs for digital pins;
- 1.8 V digital core;
- 8 V capable inputs/outputs for analog audio interfaces (capability to output 2 V<sub>RMS</sub> for SCART requirements);
- 3.3 V for stereo ADC and DAC (analog part);
- 1.8 V for stereo ADC and DAC (digital part);
- 1.8 V for IF ADC and AGC.

These voltages will be delivered by the application with an accuracy of  $\pm 5\%$ . For more information, refer to [Section 13.3: Power Supply Data](#).

Other specific DC voltages or features are provided:

- Voltage Reference and Biasing Generation (AGC, ADCs, DACs),
- Bandgap reference.

### 8.1 Standby Mode (Loop-through mode)

The STV82x7 provides a Loop-through mode configuration that bypasses IC functions via a SCART I/O pin (Full Analog Path only). In this case, only a minimum power of 200 mW is required.

In Standby mode, the digital and analog power supplies are switched off, except for pins VCC\_H, VCC33\_LS, VCC33\_SC, and VCC\_NISO which are used to maintain the SCART path with the last configuration programmed by analog matrixing (register [SCART1\\_2\\_OUTPUT\\_CTRL](#) and [SCART3\\_OUTPUT\\_CTRL](#)). When switching back to normal Full Power mode, all I<sup>2</sup>C registers are reset except for those used in Standby mode to maintain the original configuration.

In Standby mode, the I<sup>2</sup>C bus does not operate. However, the bus can still be used by other ICs since the I<sup>2</sup>C I/O pins (SDA and SCL) of the STV82x7 are forced into a high-impedance configuration.

### 8.2 Power on Reset

The following supply voltages are involved for Power on Reset for the STV82x7:

- for 1.8 V: VDD18 on pins 38, 42, 50 and 66, VCC18\_CLK1 on pin 54 and VCC18\_CLK2 on pin 57.
- for 3.3 V: VDD33\_IO1 on pin 46 and VDD33\_IO2 on pin 59.

The first condition for a valid reset is that all 1.8 V supply voltages involved have reached a minimum valid voltage of 1.7 V and that all 3.3 V supply voltages involved have reached a minimum valid voltage of 3.1 V. When this is the case and starting from this point, the reset must be maintained at a low level (<1 V) for at least 100  $\mu$ s then put to a high level.

## 9 Additional Controls and Flag

This logic contains:

- the headphone detection,
- the IRQ generation, signal to be output to the MCU,
- the I<sup>2</sup>C bus expander output pin.

### 9.1 Headphone Detection

For headphone, the  $\overline{\text{HP\_DET}}$  input can be used to automatically mute the Loudspeakers and Subwoofer outputs when the HP\_LS\_MUTE bit is set in register [HEADPHONE\\_CONFIG](#) (active low). When a headphone is detected (the  $\overline{\text{HP\_DET}}$  pin is set to 0) and the Mute function is enabled. Each change on the  $\overline{\text{HP\_DET}}$  pin generates an IRQ request to the microprocessor on the IRQ pin.

### 9.2 IRQ Generation

Four IRQs are generated by the STV82x7. On each IRQ generation, the IRQ pin is set to 1. The pending IRQ status must be read at the I<sup>2</sup>S address 81h and the acknowledge is done by writing 0 to this register.

The four availables IRQs are:

**IRQ0:** The identified TV sound standard is displayed in register [AUTOSTD\\_STATUS](#). Each change in the detected standard is flagged to the host system via hardware pin IRQ. The flag must be reset by re-programming the IRQ bit in register [AUTOSTD\\_CTRL](#) and then checking the detected standard status by reading registers [AUTOSTD\\_STATUS](#), [NICAM\\_STAT](#), and [ZWT\\_STAT](#).

**IRQ1:** This IRQ is enabled only in digital input mode. In case of I2S synchronisation loss, this IRQ is set to 1.

**IRQ2:** This IRQ is set to 1 when the device detects any change on the HP Detection pin (Headphone connection or disconnection).

**IRQ3:** On the STV82x7, same pins are used for both Headphone and Surround loudspeaker signal output. A change in the Headphone configuration (HP active or not active) will lead to a signal switch on those hardware pins. In order to ensure a smooth audio transition, the output is soft muted before the signal is switched. The IRQ3 is then set to 1 to advise the master processor that the signal has been switched and to request a HP/Srnd Output Un-Mute.

### 9.3 I<sup>2</sup>C Bus Expander

Pin BUS\_EXP can be used to control external switchable IF SAW filters or audio switches. This pin can be directly programmed by register [RESET](#).



## 10 STV82x7 Reset

All STV82x7 features are controlled via the I<sup>2</sup>C bus.

The STV82x7 can be "reset" in 2 ways:

1. By Software via the I<sup>2</sup>C bus: This clears all synchronous logic, except for the I<sup>2</sup>C bus registers.
2. By Hardware via the RESET pin: In addition to clearing all synchronous logic, the RESET input (active on the low level) resets all the I<sup>2</sup>C bus registers to the *default values* listed below.

**Table 7: RESET Default Values**

Function	Default mode
<b>Demodulation</b>	
Auto-standard	ON
Scanned Standards	M/N, B/G, I, L/L'
FM Deviation	± 125 kHz (Max.)
<b>Audio Outputs</b>	
Automatic Mute Mode	ON
Loudspeaker Source	Demodulated Sound
Loudspeaker Volume	-40 dB, differential mode, muted
Loudspeaker L/R Balance	L/R = 100%
Subwoofer	-40 dB / OFF
Headphone Source	Demodulated Sound
Headphone Automatic Detection	ON
Headphone Volume	-40 dB, differential mode, muted
Headphone L/R Balance	L/R = 100%
SCART-1 out	Demodulated Sound
SCART-2 out	SCART1 Source
SCART Volume	-5.5 dB, independent mode, muted
I <sup>2</sup> S out	OFF
<b>Audio Processing</b>	
Loudspeaker/Headphone SVC	OFF, 0 dB Reference Value
Loudspeaker Surround	OFF
Loudspeaker 5-Band Equalizer	OFF, 0 dB (Flat Band)
Loudspeaker Loudness	OFF
Headphone Bass/Treble	OFF, 0 dB (Flat Band)
Loudspeaker/Headphone Beeper	-40 dB / OFF

# 11 I<sup>2</sup>C Interface

## 11.1 I<sup>2</sup>C Address and Protocol

The STV82x7 I<sup>2</sup>C interface works in Slave mode and is fully compliant with I<sup>2</sup>C standards in Fast mode (maximum frequency of 400 kHz). Two pairs of I<sup>2</sup>C chip addresses are used to connect two STV82x7 chips to the same I<sup>2</sup>C serial bus. The device address pairs are defined by the polarity of the ADR\_SEL pin and are listed in the following table:

**Table 8: I<sup>2</sup>C Read/Write Addresses**

ADR	Write Address (W)	Read Address (R)
LOW (connected to GND1)	80h	81h
HIGH (connected to VDD1)	84h	85h

### Protocol Description

- Write Protocol

Start	W	A	Sub-address	A	Data	A	...	A	Data	A	Stop
-------	---	---	-------------	---	------	---	-----	---	------	---	------

- Read Protocol

Start	W	A	Sub-address	A	Stop	Start	R	A	Data	A	...	A	Data	N
-------	---	---	-------------	---	------	-------	---	---	------	---	-----	---	------	---

- W = Write address,
- R = Read address,
- A = Acknowledge,
- N = No acknowledge,
- Sub-address is the register address pointer; this value auto-increments for both write and read.

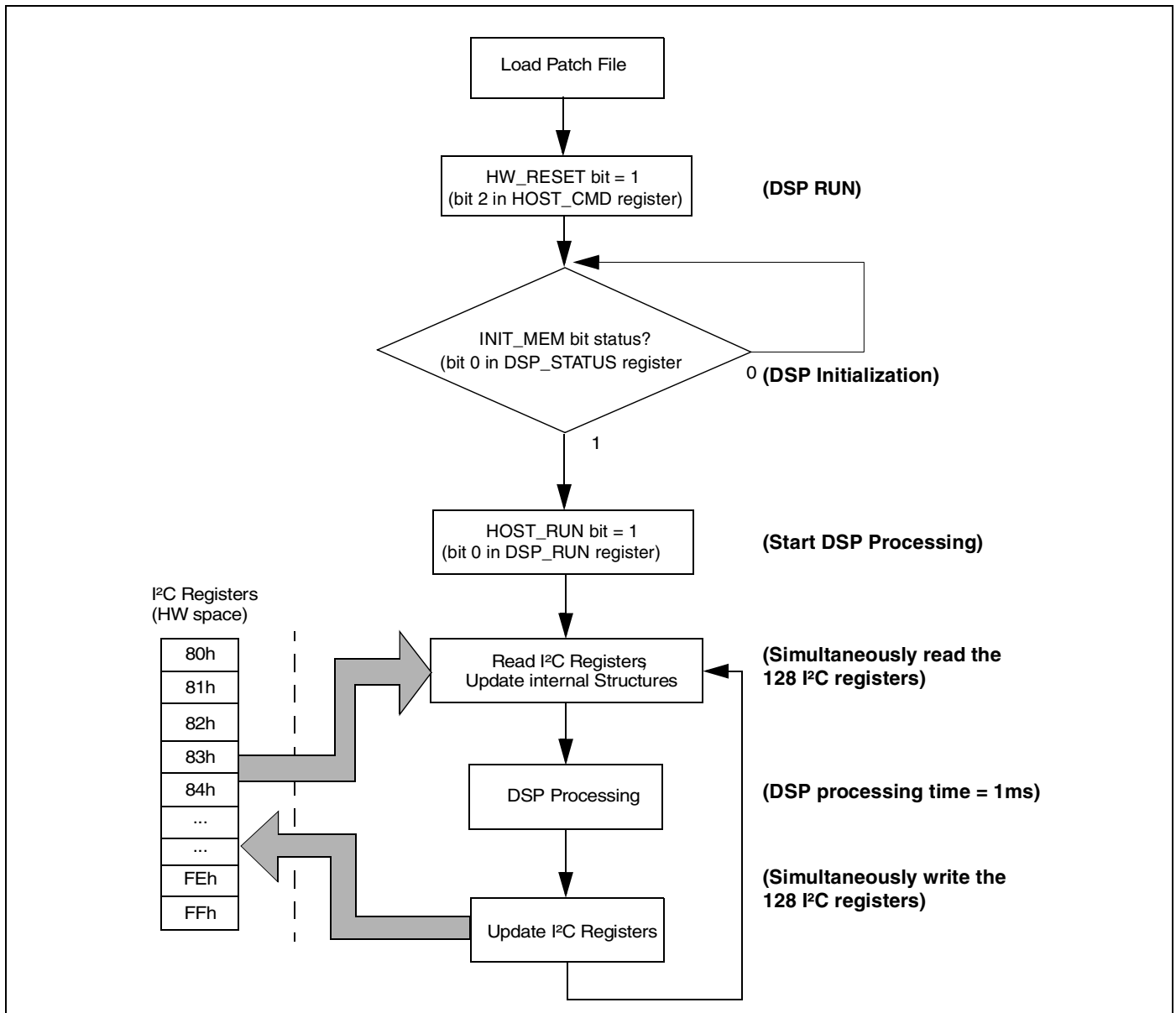
## 11.2 Start-up and Configuration Change Procedure

The DSP running loop is:

- Read IC registers and update internal structures (memory variables)
- Process sound samples
- Write I<sup>2</sup>C registers with new updated values

The step “process sound sample” duration is **1ms**. This is shown in [Figure 30](#).

**Figure 30: Simplified DSP Processing Flow**

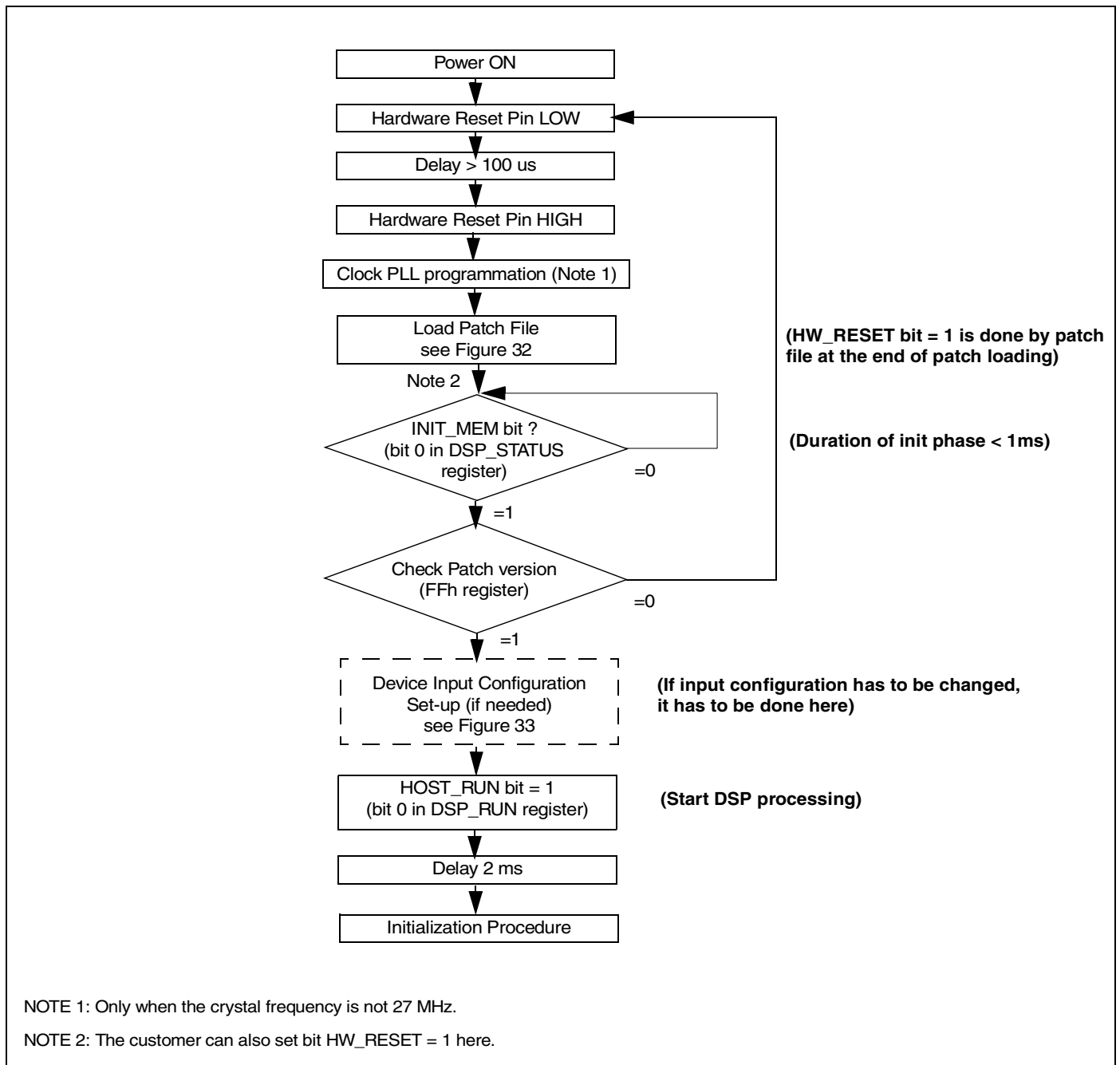


When programming I<sup>2</sup>C read/write register with addresses between 80h and FFh this flow has to be taken into account.

For example, if two different values are written in the **same** register in less than 2 ms, it is possible that the DSP doesn't see the first value (because the second value over-writes the first one during the “DSP processing” phase, before DSP can read the registers again).

In the same way, when waiting for a register value change, the software programme must wait for at least 2 ms in order to allow sufficient time for the DSP to update the register values.

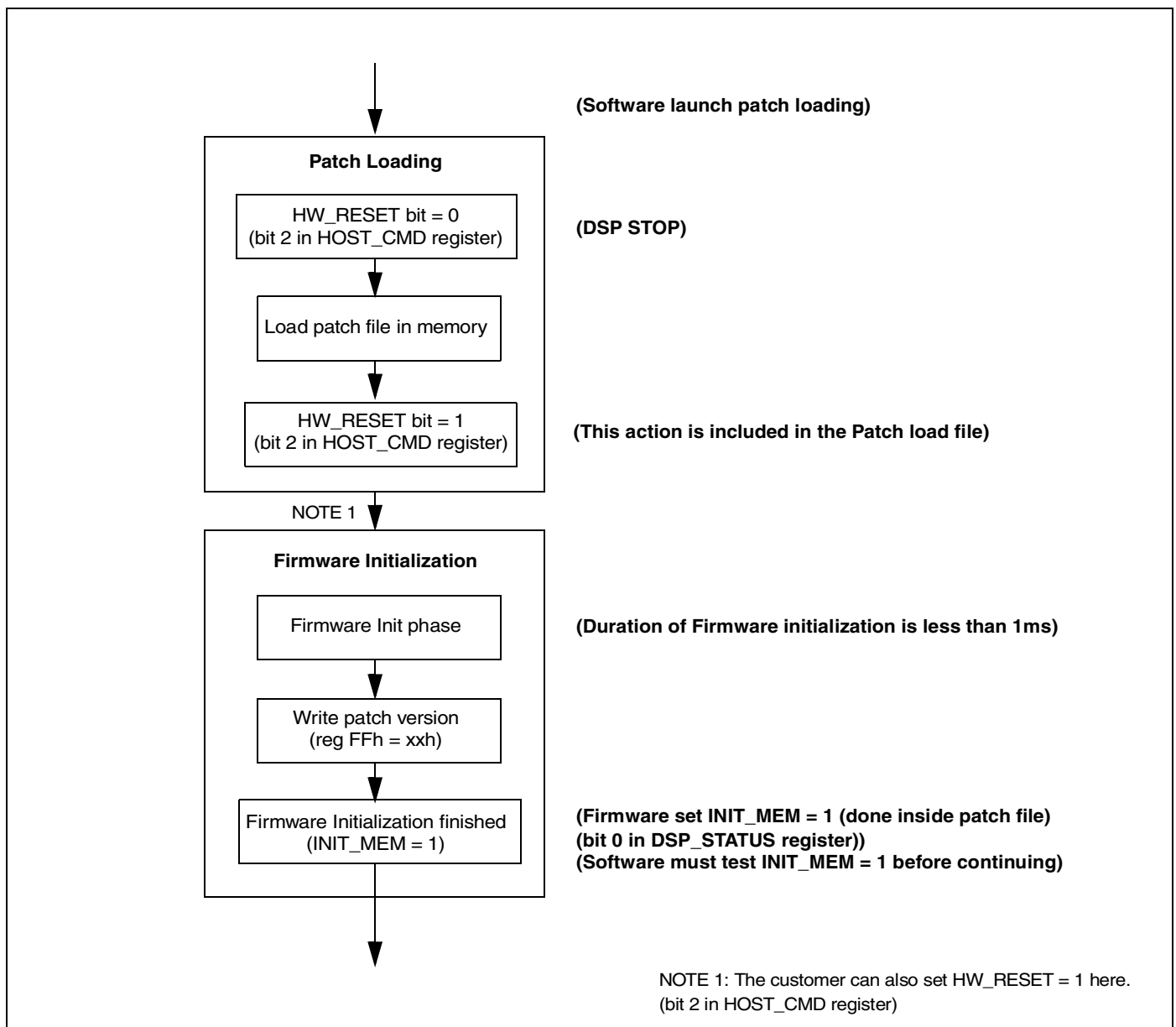
**Figure 31: Initialization Procedure at Startup**



## 11.3 Process Flow during Patch Loading and DSP Initialization

Patch loading and DSP firmware initialization are shown in [Figure 32](#)

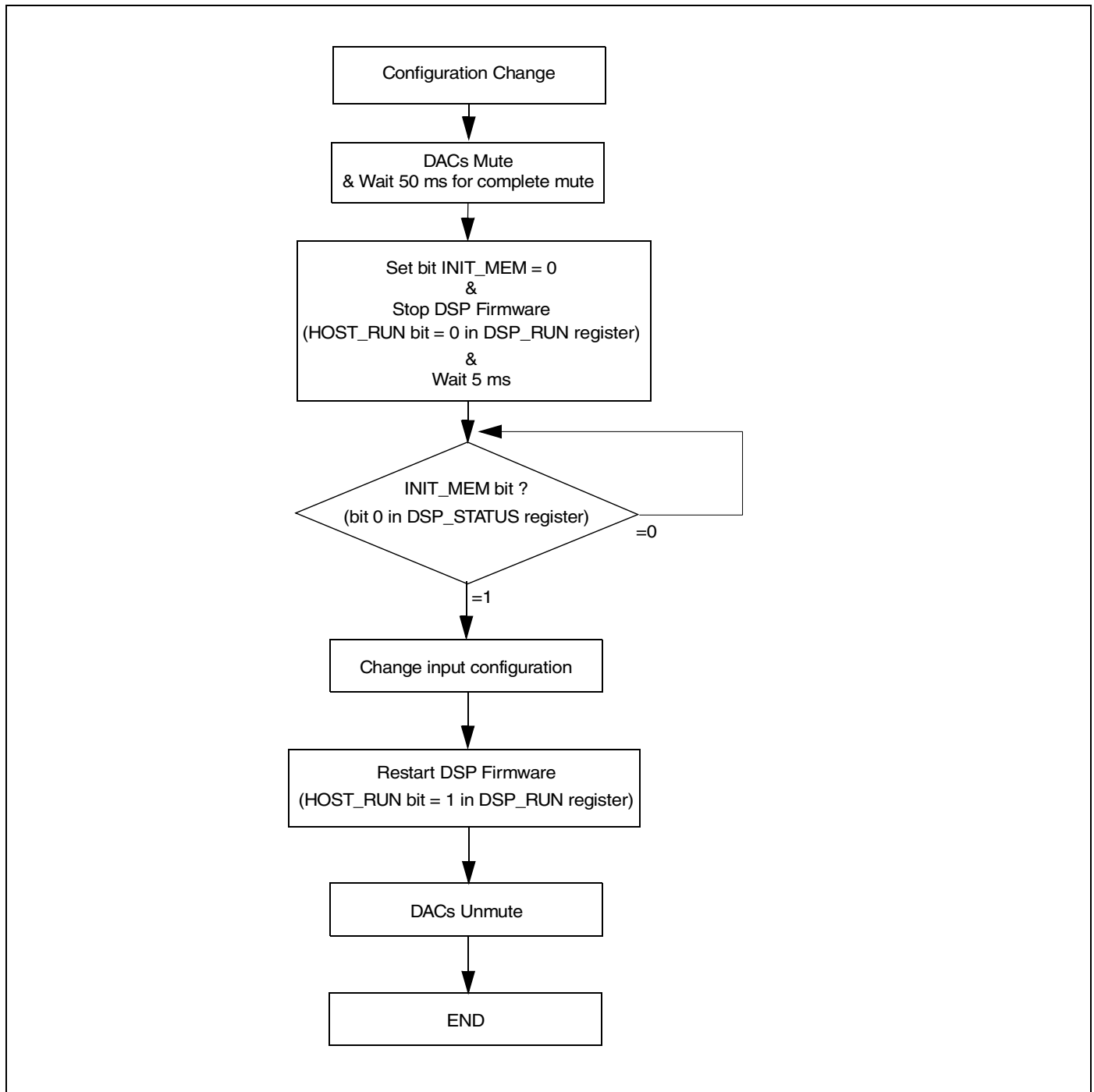
**Figure 32: Patch Loading and DSP Initialization**



## 11.4 Input Configuration Change

The input configuration change must be programmed as shown in [Figure 33](#):

**Figure 33: Input Configuration change**



## 12 Register List

*Note:* The unused bits (defined as 'Reserved') in the I<sup>2</sup>C registers must be kept to zero.

The system clock registers (from address 08h to 0Bh and from address 5Ah to 5Dh) do not need to be modified if a standard 27 MHz quartz crystal oscillator is used.

The default values of the demodulator registers (from address 0Ch to 55h) are for optimum performances and any change is not recommended, except for:

- **AGC\_GAIN** (0Fh) to adjust AGC gain for AM carrier in L/L' standard (AGC used in open loop).
- **CAROFFSET1** (22h) and **CAROFFSET2** (3Ah) to compensate IF carrier frequency with an out-of-standard offset.
- Soundlevel Prescaling **PRESCALE\_AM** (94h), **PRESCALE\_FM** (95h), **PRESCALE\_NICAM** (96h) and **PRESCALE\_SCART** (97h) to equalize demodulated or external audio signal before audio processing. Peak detector registers **PEAK\_DET\_INPUT** (9Dh), **PEAK\_DET\_L** (9Eh), **PEAK\_DET\_R** (9Fh), **PEAK\_DET\_L\_R** (A0h) can be used to measure internal sound level.

Sound source selection for each audio output channel Loudspeakers, Headphone and SCART to be done using **AUDIO\_MATRIX\_INPUT** (A2h).

In Multi-lingual mode, **AUDIO\_MATRIX\_LANGUAGE** (A4h) selects separately the language for each audio output channel.

Register **AUTOSTD\_CTRL** (8Ah) is used to select between L/L' or D/K/K1/K2/K3 standard which can be discriminated automatically. To be used also to change maximum FM deviation (125 kHz, by default) in case of wide overmodulation. **AUTOSTD\_STANDARD\_DETECT** (8Bh) and **AUTOSTD\_STEREO\_DETECT** (8Ch) to define the list of mono and stereo standards to be recognized automatically.

*Note:* ( ) used in reset value column means that the bit or the byte is read-only.  
 (S) symbol indicates that the field value is represented in signed binary format.  
 (\*) The field **AGC\_ERR[4:0]** (**AGC\_GAIN**) can be written by user if the bit **AGC\_CMD** (**AGC\_CTRL**) is set to one (by default controlled by Automatic Standard Recognition System). To be used to adjust manually the input gain of analog AGC amplifier for AM carrier (L/L').

## 12.1 I<sup>2</sup>C Register Map

By default, all I<sup>2</sup>C registers controlled by Automatic Standard Recognition System (Autostandard) are forced to Read-only mode for the user. These registers and bits are shaded in [Table 9](#).

**Table 9: List of I<sup>2</sup>C Registers (Sheet 1 of 6)**

Name	Addr.	Reset	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
<b>IC General Control</b>										
CUT_ID	00h	(0000 0001)	0	0	CUT_NUMBER[5:0]					
RESET	01h	0000 0000	BUS_EXP	I <sup>2</sup> S_OUTPUT	0	EN_STBY	0	SOFT_LRST2	SOFT_LRST1	SOFT_RST
I2S_CTRL	04h	0000 0000	SYNC_OFF	SYNC_SIGN	0	LOCK_TH[1:0]		LOCK_MODE	SYNC_CST[1:0]	
I2S_STAT	05h	(0000 0000)	0	0	0	0	0	0	LR_OFF	LOCK_FLAG
I2S_SYNC_OFFSET	06h	0000 0000	I2S_SFO[7:0]							
<b>Clocking 1</b>										
SYS_CONFIG	07h	0000 0000	I2S_CH_NB[1:0]		INPUT_FREQ[3:0]			INPUT_CONFIG[1:0]		
FS1_DIV	08h	0001 0010	EN_PROG	0	NDIV1[1:0]		0	SDIV1[2:0]		
FS1_MD	09h	0001 0001	0	0	0	MD1[4:0]				
FS1_PE_H	0Ah	0011 0110	PE_H1[7:0]							
FS1_PE_L	0Bh	0000 0000	PE_L1[7:0]							
<b>Demodulator</b>										
DEMOD_CTRL	0Ch	0000 0110	0	0	FAR_MODE	GAP_MODE	AM_SEL	DEMOD_MODE[2:0]		
DEMOD_STAT	0Dh	(0000 0000)	0	0	0	QPSK_LK	FM2_CAR	FM2_SQ	FM1_CAR	FM1_SQ
AGC_CTRL	0Eh	0001 0001	AGC_CMD	0	0	AGC_REF[2:0]			AGC_CST[1:0]	
AGC_GAIN	0Fh	(0000 0000)	0	AGC_ERR[4:0]					SIG_OVER	SIG_UNDER
DC_ERR_IF	10h	(0000 0000)	DC_ERR[7:0]							
<b>Demodulator Channel 1</b>										
CARFQ1H	12h	0011 1110	CARFQ1[23:16]							
CARFQ1M	13h	1000 0000	CARFQ1[15:8]							
CARFQ1L	14h	0000 0000	CARFQ1[7:0]							
FIR1C0	15h	0000 0000	FIR1C0[7:0] (S)							
FIR1C1	16h	1111 1110	FIR1C1[7:0] (S)							
FIR1C2	17h	1111 1100	FIR1C2[7:0] (S)							
FIR1C3	18h	1111 1101	FIR1C3[7:0] (S)							
FIR1C4	19h	0000 0010	FIR1C4[7:0] (S)							
FIR1C5	1Ah	0000 1101	FIR1C5[7:0] (S)							
FIR1C6	1Bh	0001 1000	FIR1C6[7:0]6 (S)							
FIR1C7	1Ch	0001 1111	FIR1C7[7:0] (S)							
ACOEFF1	1Dh	0010 0011	ACOEFF1[7:0]							
BCOEFF1	1Eh	0001 0010	BCOEFF1[7:0]							
CRF1	1Fh	(0000 0000)	CRF1[7:0] (S)							
CETH1	20h	0010 0000	CETH1[7:0]							



Table 9: List of I<sup>2</sup>C Registers (Sheet 2 of 6)

Name	Addr.	Reset	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
SQTH1	21h	0011 1100	SQTH1[7:0]							
CAROFFSET1	22h	0000 0000	CAROFFSET1[7:0] (S)							
<b>Demodulator Channel 2</b>										
IAGCR	25h	1000 1000	IAGC_REF[7:0]							
IAGCC	26h	0000 0011	IAGC_OFF	FAR_FLT_EN	MONO_FLT_EN	BG_SEL	MONO_PROG	IAGC_CST[2:0]		
IAGCS	27h	(0000 0000)	IAGC_CTRL[7:0]							
CARFQ2H	28h	0100 0100	CARFQ2[23:16]							
CARFQ2M	29h	0100 0000	CARFQ2[15:8]							
CARFQ2L	2Ah	0000 0000	CARFQ2[7:0]							
FIR2C0	2Bh	0000 0000	FIR2C0[7:0] (S)							
FIR2C1	2Ch	0000 0000	FIR2C1[7:0] (S)							
FIR2C2	2Dh	0000 0000	FIR2C2[7:0] (S)							
FIR2C3	2Eh	0000 0000	FIR2C3[7:0] (S)							
FIR2C4	2Fh	1111 1111	FIR2C4[7:0] (S)							
FIR2C5	30h	0000 0100	FIR2C5[7:0] (S)							
FIR2C6	31h	0001 0100	FIR2C6[7:0] (S)							
FIR2C7	32h	0010 0101	FIR2C7[7:0] (S)							
ACOEFF2	33h	1001 0000	ACOEFF2[7:0]							
BCOEFF2	34h	1010 1100	BCOEFF2[7:0]							
SCOEFF	35h	0001 1100	SCOEFF[7:0]							
SRF	36h	(0000 0000)	SRF[7:0] (S)							
CRF2	37h	(0000 0000)	CRF2[7:0] (S)							
CAROFFSET2	3Ah	0000 0000	CAROFFSET2[7:0] (S)							
<b>NICAM</b>										
NICAM_CTRL	3Dh	0000 0000	0	0	0	0	0	DIF_POL	ECT	MAE
NICAM_BER	3Eh	(0000 0000)	ERROR[7:0]							
NICAM_STAT	3Fh	(0000 0000)	NIC_DET	F_MUTE	LOA	CBI[3:0]			NIC_MUTE	
<b>Stereo FM</b>										
ZWT_CTRL	40h	0011 0001	LRST_TONE_OFF	STD_MODE	THRESH[3:0]			TSCCTRL[1:0]		
ZWT_TIME	41h	0000 0100	0	0	0	0	ZWT_TIME[2:0]			
ZWT_STAT	42h	(0000 0000)	0	0	0	0	ZW_STAT_RDY	ZW_DET	ZW_ST	ZW_DM
<b>Analog Control</b>										
ADC_CTRL	56h	0000 1000	I2S_DATA0_CTRL[1:0]		0	0	ADC_POWER_UP	ADC_INPUT_SEL[2:0]		
SCART1_2_OUTPUT_CTRL	57h	1010 1000	SC2_MUTE	SC2_OUTPUT_SEL[2:0]			SC1_MUTE	SC1_OUTPUT_SEL[2:0]		
SCART3_OUTPUT_CTRL	58h	0000 1011	0	0	0	0	SC3_MUTE	SC3_OUTPUT_SEL[2:0]		
<b>Clocking 2</b>										
FS2_DIV	5Ah	0001 0001	0	NDIV2[1:0]			0	SDIV2[2:0]		

Table 9: List of I<sup>2</sup>C Registers (Sheet 3 of 6)

Name	Addr.	Reset	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0	
FS2_MD	5Bh	0001 0001	0	0	0	MD2[4:0]					
FS2_PE_H	5Ch	0101 1100	PE_H2[7:0]								
FS2_PE_L	5Dh	0010 1001	PE_L2[7:0]								
<b>DSP Control</b>											
HOST_CMD	80h	0000 0000	IT_IN_DSP	0	0	0	0	HW_RESET	0	0	
IRQ_STATUS	81h	0000 0000	0	0	0	0	IRQ3 (HP/Srmd unmute ready)	IRQ2 (HP detected)	IRQ1 (I2S sync lost)	IRQ0 (autostd)	
SOFT_VERSION	82h	(0000 0002)	SOFT_VERSION[7:0]								
ONCHIP_ALGOS	83h	(0000 0000)	0	PRO_LOGIC_SELECT	NICAM	I2S_INPUT	TRUBASS	TRU SURROUND	PRO_LOGIC	MULTICHANNEL	
DSP_STATUS	84h	0000 0000	0	0	0	0	0	0	0	INIT_MEM	
DSP_RUN	85h	0000 0000	0	0	0	0	0	0	HOST_NO_INIT	HOST_RUN	
I2S_IN_CONFIG	86h	1000 1110	LOCK_MODE_EN	0	SYNC	LRCLK_START	LRCLK_POLARITY	SCLK_POLARITY	DATA_CFG	I2S_MODE	
AV_DELAY	89h	0000 0000	DELAY_TIME[6:0]								DELAY_ON
<b>Automatic Standard Recognition System</b>											
AUTOSTD_CTRL	8Ah	0000 0001	0	0	0	FORCE_SQUELCH	SINGLE_SHOT	DK_DEV[1:0]		LDK_SW	
AUTOSTD_STANDARD_DETECT	8Bh	0010 1111	0	NICAM_C4_OFF	NICAM_GAP_MODE	NICAM_MONO_IN	LDK_SCK	I_SCK	BG_SCK	MN_SCK	
AUTOSTD_STEREO_DETECT	8Ch	0001 1111	LDK_ZWT3	LDK_ZWT2	LDK_SWT1	LDK_NICAM	I_NICAM	BG_ZWT	BG_NICAM	MN_ZWT	
AUTOSTD_TIMERS	8Dh	1010 0100	FM_TIME[1:0]		NICAM_TIME[2:0]			ZWEITON_TIME[2:0]			
AUTOSTD_STATUS	8Eh	(0000 0000)	STEREO_ID	STEREO_OK	MONO_OK	AUTOSTD_ON	STEREO_SID[1:0]		MONO_SID[1:0]		
<b>Audio Preprocessing &amp; Selection</b>											
DC_REMOVAL_INPUT	90h	0000 0111	0	0	0	0	0	DC_SCART	DC_NICAM	DC_DEMOD	
DC_REMOVAL_L	91h	(0000 0000)	DC_REMOVAL_L[7:0] (S)								
DC_REMOVAL_R	92h	(0000 0000)	DC_REMOVAL_R[7:0] (S)								
PRESCALE_SELECT	93h	0000 0000	0	0	0	0	0	0	0	AM_FM_SELECT	
PRESCALE_AM	94h	0000 0000	0	PRESCALE_AM[6:0] (S)							
PRESCALE_FM	95h	0000 1100	0	PRESCALE_FM[6:0] (S)							
PRESCALE_NICAM	96h	0001 1010	0	PRESCALE_NICAM[6:0] (S)							
PRESCALE_SCART	97h	0000 0000	0	0	PRESCALE_SCART[5:0] (S)						
PRESCALE_I2S_0	98h	0000 0000	0	0	PRESCALE_I2S_0[5:0] (S)						
PRESCALE_I2S_1	99h	0000 0000	0	0	PRESCALE_I2S_1[5:0] (S)						
PRESCALE_I2S_2	9Ah	0000 0000	0	0	PRESCALE_I2S_2[5:0] (S)						
DEEMPHASIS_DEMATRIX	9Bh	0000 0000	0	0	NICAM_DEMATRIX	NICAM_DEEMPH_BYPASS	FM_DEMATRIX[1:0]		FM_DEEMPH_BYPASS	FM_DEEMPH_SW	
PEAK_DET_INPUT	9Dh	0000 0000	PEAK_LOCATION	0	PEAK_L_R_RANGE				PEAK_DET_INPUT[1:0]		

Table 9: List of I<sup>2</sup>C Registers (Sheet 4 of 6)

Name	Addr.	Reset	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0	
PEAK_DET_L	9Eh	0(0000 0000)	OVERLOAD_L [7:0]	PEAK_L[6:0]							
PEAK_DET_R	9Fh	0(0000 0000)	OVERLOAD_R [7:0]	PEAK_R[6:0]							
PEAK_DET_L_R	A0h	0(0000 0000)	OVERLOAD_L_R [7:0]	PEAK_L_R[6:0]							

**Matrixing**

AUDIO_MATRIX_INPUT	A2h	0000 0000	0	0	0	0	0	SCART_INPUT_SOURCE	HP_INPUT_SOURCE	LS_INPUT_SOURCE
AUDIO_MATRIX_CONFIG	A3h	0000 0000	0	0	0	SCART_MATRIX	DEMOM_MATRIX[3:0]			
AUDIO_MATRIX_LANGUAGE	A4h	0000 0000	MUTE_STEREO	MUTE_ALL	SCART_LANGUAGE[1:0]		HP_LANGUAGE[1:0]		LS_LANGUAGE[1:0]	
DOWNMIX_IN_MODE	A6h	0000 0010	0	0	0	0	LFE_IN	MIX_IN_MODE[2:0]		
DOWNMIX_OUT_MODE	A7h	0100 1010	0	HP_MODE[1:0]		SCART_MODE[1:0]		MIX_OUT_MODE[2:0]		
DOWNMIX_DUAL_MODE	A8h	0000 0000	0	DUAL_ON	LS_DUAL_SELECT[1:0]		SCART_DUAL_SELECT [1:0]		HP_DUAL_SELECT[1:0]	
DOWNMIX_CONFIG	A9h	0000 0001	0	0	SRND_FACTOR[1:0]		CENTER_FACTOR[1:0]		LR_UPMIX	NORMALIZE

**Audio Processing**

PRO_LOGIC2_CONTROL	AAh	0011 1010	PL2_LFE	PL2_OUTPUT_DOWNMIX[2:0]			PL2_MODES[2:0]			PL2_ACTIVE
PCM_SRND_DELAY	ABh	0000 0000	0	0	0	SNRD_DELAY[4:0]				
PCM_CENTER_DELAY	ACH	0000 0000	0	0	0	0	CENTER_DELAY[3:0]			
PRO_LOGIC2_CONFIG	ADh	0000 0000	0	0	0	PL2_SRND_FILTER	PL2_RS_POLARITY	PL2_PANORAMA	PL2_AUTO BALANCE	
PRO_LOGIC2_DIMENSION	Aeh	0000 0000	0	PL2_C_WIDTH			0	PL2_DIMENSION		
PRO_LOGIC2_LEVEL	Afh	0000 0000	PL2_LEVEL							
NOISE_GENERATOR	B0h	0000 0000	10_DB_ATTENUATE	SRIGHT_NOISE	SLEFT_NOISE	SUB_NOISE	CENTER_NOISE	RIGHT_NOISE	LEFT_NOISE	NOISE_ON
TRUSRND_CONTROL	B1h	0000 0000	0	TRUSRND_MONO_SRND	TRUSRND_INPUT_MODE[3:0]				TRUSRND_MODE	TRUSRND_ON
TRUSRND_INPUT_GAIN	B6h	0000 0000	TRUSRND_INPUT_GAIN[7:0]							
TRUSRND_HP_DCL	B7h	0000 0000	0	0	0	0	0	DIALOG_CLARITY_ON	HEADPHONE_ON	0
TRUSRND_DC_ELEVATION	B8h	0000 1100	TRUSRND_DC_ELEVATION[7:0]							
TRUBASS_LS_CONTROL	BAh	0000 0110	0	0	0	TRUBASS_LS_SIZE[3:0]				TRUBASS_LS_ON
TRUBASS_LS_LEVEL	BBh	00001 1001	TRUBASS_LS_LEVEL[7:0]							
TRUBASS_HP_CONTROL	BCh	0000 0110	0	0	0	TRUBASS_HP_SIZE[3:0]				TRUBASS_HP_ON
TRUBASS_HP_LEVEL	BDh	0000 1001	TRUBASS_HP_LEVEL[7:0]							
SVC_LS_CONTROL	BEh	0000 0010	0	0	0	0	SVC_LS_INPUT[1:0]		SVC_LS_AMP	SVC_LS_ON
SVC_LS_TIME_TH	Bfh	1001 1000	SVC_LS_TIME[2:0]				SVC_LS_THRESHOLD[4:0] (S)			
SVC_HP_CONTROL	C0h	0000 0010	0	0	0	0	0	0	SVC_LHP_AMP	SVC_HP_ON
SVC_HP_TIME_TH	C1h	1001 1000	SVC_HP_TIME[2:0]				SVC_HP_THRESHOLD[4:0] (S)			

Table 9: List of I<sup>2</sup>C Registers (Sheet 5 of 6)

Name	Addr.	Reset	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
SVC_LS_GAIN	C2h	0000 0000	0	0	0	SVC_LS_MAKE_UP_GAIN[4:0]				
SVC_HP_GAIN	C3h	0000 0000	0	0	0	SVC_HP_MAKE_UP_GAIN[4:0]				
STSRND_CONTROL	C4h	0000 0000						STSRND_STEREO	STSRND_MODE	STSRND_ON
STSRND_FREQ	C5h	0001 0101	0	0	STSRND_BASS[1:0]		STSRND_MEDIUM[1:0]		STSRND_TREBLE[1:0]	
STSRND_LEVEL	C6h	1000 0000	STSRND_GAIN[7:0]							
OMNISURROUND_CONTROL	C7h	0000 0000	ST_VOICE			OMNISURROUND_INPUT_MODE				OMNISURROUND_ON
ST_DYNAMIC_BASS	C8h	0000 0000	BASS_LEVEL					BASS_FREQ		DYN_BASS_ON
LS_EQ_BT_CTRL	C9h	0000 0000	0	0	0	0	0	0	LS_EQ_BT_SW	LS_EQ_ON
LS_EQ_BAND1	CAh	0000 0000	EQ_BAND1[7:0] (S)							
LS_EQ_BAND2	CBh	0000 0000	EQ_BAND2[7:0] (S)							
LS_EQ_BAND3	CCh	0000 0000	EQ_BAND3[7:0] (S)							
LS_EQ_BAND4	CDh	0000 0000	EQ_BAND4[7:0] (S)							
LS_EQ_BAND5	CEh	0000 0000	EQ_BAND5[7:0] (S)							
LS_BASS_GAIN	CFh	0000 0000	LS_BASS[7:0] (S)							
LS_TREBLE_GAIN	D0h	0000 0000	LS_TREBLE[7:0] (S)							
HP_BT_CONTROL	D1h	0000 0000	0	0	0	0	0	0	0	HP_BT_ON
HP_BASS_GAIN	D2h	0000 0000	HP_BASS[7:0] (S)							
HP_TREBLE_GAIN	D3h	0000 0000	HP_TREBLE[7:0] (S)							
OUTPUT_BASS_MNGT	D4h	0000 0000	BASS_MANAGE_ON	0	SUB_ACTIVE	GAIN_SWITCH	0	OCFG_NUM[2:0]		
LS_LOUDNESS	D5h	0000 0100	0	LS_LOUD_THRESHOLD[2:0]			LS_LOUD_GAIN_HR[2:0]			LS_LOUD_ON
HP_LOUDNESS	D6h	0000 0100	0	HP_LOUD_THRESHOLD[2:0]			HP_LOUD_GAIN_HR[2:0]			HP_LOUD_ON

**Volume**

VOLUME_MODES	D7h	1100 0111	ANTCLIP_HP_VOL_CLAMP	ANTCLIP_LS_VOL_CLAMP	0	0	SCART_VOLUME_MODE	SRND_VOLUME_MODE	HP_VOLUME_MODE	LS_VOLUME_MODE
LS_L_VOLUME_MSB	D8h	1001 1000	LS_L_VOLUME_MSB[7:0]							
LS_L_VOLUME_LSB	D9h	0000 0000	0	0	0	0	0	0	LS_L_VOLUME_LSB[1:0]	
LS_R_VOLUME_MSB	DAh	0000 0000	LS_R_VOLUME_MSB[7:0]							
LS_R_VOLUME_LSB	DBh	0000 0000	0	0	0	0	0	0	LS_R_VOLUME_LSB[1:0]	
LS_C_VOLUME_MSB	DCh	1001 1000	LS_C_VOLUME_MSB[7:0]							
LS_C_VOLUME_LSB	DDh	0000 0000	0	0	0	0	0	0	LS_C_VOLUME_LSB[1:0]	
LS_SUB_VOLUME_MSB	DEh	1001 1000	LS_SUB_VOLUME_MSB[7:0]							
LS_SUB_VOLUME_LSB	DFh	0000 0000	0	0	0	0	0	0	LS_SUB_VOLUME_LSB[1:0]	
LS_SL_VOLUME_MSB	E0h	1001 1000	LS_SL_VOLUME_MSB[7:0]							
LS_SL_VOLUME_LSB	E1h	0000 0000	0	0	0	0	0	0	LS_SL_VOLUME_LSB[1:0]	
LS_SR_VOLUME_MSB	E2h	0000 0000	LS_SR_VOLUME_MSB[7:0]							
LS_SR_VOLUME_LSB	E3h	0000 0000	0	0	0	0	0	0	LS_SR_VOLUME_LSB[1:0]	

Table 9: List of I<sup>2</sup>C Registers (Sheet 6 of 6)

Name	Addr.	Reset	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
LS_MASTER_VOLUME_MSB	E4h	1110 1000	LS_MASTER_VOLUME_MSB[7:0]							
LS_MASTER_VOLUME_LSB	E5h	0000 0000	0	0	0	0	0	0	LS_MASTER_VOLUME_LSB[1:0]	
HP_L_VOLUME_MSB	E6h	1001 1000	HP_L_VOLUME_MSB[7:0]							
HP_L_VOLUME_LSB	E7h	0000 0000	0	0	0	0	0	0	HP_L_VOLUME_LSB[1:0]	
HP_R_VOLUME_MSB	E8h	0000 0000	HP_R_VOLUME_MSB[7:0]							
HP_R_VOLUME_LSB	E9h	0000 0000	0	0	0	0	0	0	HP_R_VOLUME_LSB[1:0]	
SCART_L_VOLUME_MSB	EAh	1101 1101	SCART_L_VOLUME_MSB[7:0]							
SCART_L_VOLUME_LSB	EBh	0000 0000	0	0	0	0	0	0	SCART_L_VOLUME_LSB[1:0]	
SCART_R_VOLUME_MSB	ECh	1101 1101	SCART_R_VOLUME_MSB[7:0]							
SCART_R_VOLUME_LSB	EDh	0000 0000	0	0	0	0	0	0	SCART_R_VOLUME_LSB[1:0]	

**Beeper**

BEEPER_ON	Eeh	0000 0000	0	0	0	0	0	0	0	BEEPER_ON
BEEPER_MODE	Efh	0000 0011	0	0	0	BEEPER_DURATION[1:0]		BEEPER_PULSE	BEEPER_PATH[1:0]	
BEEPER_FREQ_VOL	F0h	0111 0000	BEEPER_FREQ[2:0]			BEEPER_VOLUME[4:0]				

**Mute**

MUTE_DIGITAL	F1h	1001 1111	AUTOSTD_MUTE_ON	0	0	SCART_D_MUTE	SRND_HP_D_MUTE	SUB_D_MUTE	C_D_MUTE	LS_D_MUTE
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**S/PDIF**

S/PDIF_OUT_CONFIG	F2h	0000 0100	0	0	0	0	0	SPDIF_OUT_MUTE	S/PDIF_OUT_SELECT[2:0]	
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**Headphone Configuration**

HEADPHONE_CONFIG	F3h	0000 001(0)	0	0	0	0	HP_FORCE	HP_LS_MUTE	HP_DET_ACTIVE	HP_DETECTED
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**DAC Control**

DAC_CONTROL	F4h	0001 1111	0	0	S/PDIF_MUX	DAC_SCART_MUTE	DAC_SHP_MUTE	DAC_CSUB_MUTE	DAC_LSLR_MUTE	POWER_UP
DAC_SW_CHANNELS	F5h	0000 0000	SUR_HP_SW		C_SUB_SW		LS_L_R_SW		SCART_SW	
SPDIF_SW_CHANNELS	F6h	0000 0000	0	0	0	0	0	0	SPFI_SW	
SPDIF_CHANNEL_STATUS	F9h	0000 0000	CHANNEL_STATUS		EMPHASIS			COPYRIGHT	NON_AUDIO	PRO_CON

**AutoStandard Coefficients Settings**

AUTOSTD_COEFF_CTRL	FBh	0000 0001	0	0	0	0	0	0	AUTOSTD_COEFF_CTRL[1:0]	
AUTOSTD_COEFF_INDEX_MSB	FCh	0000 0000	0	0	0	0	0	0	0	AUTOSTD_COEFF_INDEX_MSB
AUTOSTD_COEFF_INDEX_LSB	FDh	0000 0000	AUTOSTD_COEFF_INDEX_LSB[7:0]							
AUTOSTD_COEFF_VALUE	FEh	0000 0000	AUTOSTD_COEFF_VALUE[7:0]							
PATCH_VERSION	FFh	0000 0000	PATCH_VERSION[7:0]							

## 12.2 STV82x7 General Control Registers

### CUT\_ID

### Version Identification

Address: 00h

Type: R

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	CUT_NUMBER[5:0]					

Bit Name	Reset	Function
Bits[7:6]	00	Reserved
CUT_NUMBER[5:0]	000001	Dice Version Identification

### RESET

### Software Reset Register

Address: 01h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
BUS_EXP	I <sup>2</sup> S_OUTPUT	0	EN_STBY	0	SOFT_LRST2	SOFT_LRST1	SOFT_RST

#### Description

The built-in Automatic Standard Recognition System (Autostandard) can be disabled. In this case, the Software Reset function (bits SOFT\_LRST1 and SOFT\_LRST2) can be used to implement the Automatic Standard Recognition by I<sup>2</sup>C Software. This is not required if the built-in Automatic Standard Recognition System function is used (default).

Bit Name	Reset	Function
BUS_EXP	0	Static control by I2C of hardware pin BUS_EXP
I <sup>2</sup> S_OUTPUT	0	0: I <sup>2</sup> S Input (I2S_DATA0 , I2S_SCLK, I2S_LR_CLK, I2S_PCM_CLK in input mode) 1: I <sup>2</sup> S Output (I2S_DATA0 , I2S_SCLK, I2S_LR_CLK, I2S_PCM_CLK in output mode, 512 x Fs will be provided on the I2S_PCM_CLK pin)
Bit[5]	0	Reserved.
EN_STBY	0	<b>Standby mode enabling</b> 0: Normal mode 1: To lock the digital signals before to settle the device in standby mode
Bit 3	0	Reserved.
SOFT_LRST2	0	Softreset (active high) of Channel 2 detectors only.
SOFT_LRST1	0	Softreset (active high) of Channel 1 detectors only.
SOFT_RST	0	General softreset (active high) to reset all hardware registers except for I <sup>2</sup> C data.

## I2S\_CTRL

I<sup>2</sup>S Synchronization Control Register

Address: 04h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
SYNC_OFF	SYNC_SIGN	0	LOCK_TH[1:0]		LOCK_MODE		SYNC_CST[1:0]

Bit Name	Reset	Function
SYNC_OFF	0	Open the loop of synchronization - External PCM clock is used internally and must be equal to $512 \times f_{SOUT}$
SYNC_SIGN	0	<b>Sign of the loop reversion</b> (to be used in case of gain inversion of the Frequency Synthesizer)
Bit 5	0	Reserved
LOCK_TH[1:0]	00	<b>Lock Detector Threshold Programming</b> 00: $\pm 1$ CLK period error of accumulation 01: $\pm 2$ CLK period error of accumulation 10: $\pm 4$ CLK period error of accumulation 11: $\pm 8$ CLK period error of accumulation
LOCK_MODE	0	<b>Lock Detector Mode</b> 0: Lock when accumulation error within lock threshold and LR detected (period counter not saturated) 1: Lock when only accumulation error within lock threshold. Don't care about the LR detection
SYNC_CST[1:0]	01	<b>Synchronization Time Constant</b> Defines the measurement period of LR 00: Half period measured (lowest accuracy) 01: One full period measured 10: Two full periods measured 11: Four full periods measured (highest accuracy)

## I2S\_STAT

I<sup>2</sup>S Synchronization Status Register

Address: 05h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	0	LR_OFF	LOCK_FLAG

Bit Name	Reset	Function
Bits[7:2]	0	Reserved.
LR_OFF	0	<b>LR Signal Detection</b> 0: LR signal detected and correct 1: Missing LR pulses detected
LOCK_FLAG	0	Lock Flag allowing unmute of Audio Output

**I2S\_SYNC\_OFFSET****I<sup>2</sup>S Synchronization Offset Frequency Register**

Address: 06h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
I2S_SFO[7:0]							

Bit Name	Reset	Function
I2S_SFO[7:0]	0000 0000	I <sup>2</sup> S synchronization frequency offset ( $\pm 450$ ppm full scale)

**12.3 Clocking 1**

A low-jitter PLL Clock is integrated and can be fully reprogrammed using the registers described below. By default, the programming is defined for a 27-MHz quartz crystal frequency, which is the frequency recommended for reducing potential RF interference in the application. However, if necessary, the PLL Clock can be re-programmed for other quartz crystal frequencies within a range from 23 to 30 MHz. Other quartz crystal frequencies can be programmed on your demand.

*Note:* A Crystal Frequency change is compatible with other default I<sup>2</sup>C programming including the built-in Automatic Standard Recognition System.

**SYS\_CONFIG****System Configuration Control Register**

Address: 07h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
I2S_CH_NB[1:0]		INPUT_FREQ[3:0]			INPUT_CONFIG[1:0]		

Bit Name	Reset	Function
I2S_CH_NB[1:0]	00	<b>Number of I2S channels input</b> 00: N/A 01: 2 channels 10: 4 channels 11: 6 channels
INPUT_FREQ[3:0]	0000	<b>I2S Input frequency</b> 0000 : 32 kHz 0001: 44.1 kHz 0010: 48 kHz 0011: 8 kHz (I2S input, 2 channels only) 0100 : 11.025 kHz (I2S input, 2 channels only) 0101 : 12 kHz (I2S input, 2 channels only) 0110 : 16 kHz (I2S input, 2 channels only) 0111 : 22.05 kHz (I2S input, 2 channels only) 1000 : 24 kHz (I2S input, 2 channels only)



Bit Name	Reset	Function
INPUT_CONFIG[1:0]	0	<b>Input stream to process</b> 0 : SIF & SCART input (32 kHz) 1 : SCART input only (48 kHz) 2 : I2S input only

**FS1\_DIV****FS1 I/O Divider Programming Register**

Address: 08h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
EN_PROG	0	NDIV1[1:0]		0	SDIV1[2:0]		

Bit Name	Reset	Function
EN_PROG	0	FS1 programming enable 0: FS1 I2C registers programming ignored by system - FS1 pre-programmed automatically by SYS-CONFIG register (normal use with standard quartz of 27 MHz) 1: FS1 I2C registers programming used by system - FS1 pre-programming by SYS-CONFIG deactivated (to be used in case of no standard quartz, different from 27 MHz)
Bit 6	0	Reserved.
NDIV1[1:0]	01	FS1 Input clock divider selection
Bit 3	0	Reserved.
SDIV1[2:0]	010	FS1 Output clock divider selection

**FS1\_MD****FS1 Coarse Selection Register**

Address: 09h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	MD1[4:0]				

Bit Name	Reset	Function
Bits[7:5]	000	Reserved.
MD1[4:0]	10001	FS1 Coarse Selection

## FS1\_PE\_H

## FS1 Fine Selection Register (MSBs)

Address: 0Ah

Type: R/W

Bit 7                      Bit 6                      Bit 5                      Bit 4                      Bit 3                      Bit 2                      Bit 1                      Bit 0

PE\_H1[7:0]

Bit Name	Reset	Function
PE_H1[7:0]	0011 0110	FS1 Fine Selection (MSBs)

## FS1\_PE\_L

## FS1 Fine Selection Register (LSBs)

Address: 0Bh

Type: R/W

Bit 7                      Bit 6                      Bit 5                      Bit 4                      Bit 3                      Bit 2                      Bit 1                      Bit 0

PE\_L1[7:0]

Bit Name	Reset	Function
PE_L1[7:0]	0000 0000	FS1 Fine Selection (LSBs)

## 12.4 Demodulator

## DEMOD\_CTRL

## Demodulator Control Register

Address: 0Ch

Type: R/W

Bit 7                      Bit 6                      Bit 5                      Bit 4                      Bit 3                      Bit 2                      Bit 1                      Bit 0

0

0

FAR\_MODE

GAP\_MODE

AM\_SEL

DEMOD\_MODE[2:0]

Bit Name	Reset	Function
bit [7:6]	000	Reserved
FAR_MODE	0	1: Farrow and Mono filter for NICAM active
GAP_MODE	0	<b>Defines the clock gapping mode of the demodulator</b> 0: (default), the FS1 freq is controlled by stl-error (clock-pll mode) to align the instantaneous value of the internal clock with respect to the received NICAM clock 1: the FS1 freq is fixed and the mean value of the internal clock is aligned by variable gapping (src-error) with respect to the received NICAM clock

Bit Name	Reset	Function																											
AM_SEL	0	<b>Demodulator Configuration Select</b> 0: FM configuration of demodulator (Default) 1: AM configuration of demodulator																											
DEMOD_MODE[2:0]	110	<b>Demodulator Mode Select</b>  <table border="0"> <thead> <tr> <th></th> <th><u>CH1 FM</u></th> <th><u>CH2 FM/QPSK</u></th> </tr> </thead> <tbody> <tr> <td>000:</td> <td>Normal</td> <td>FM Normal</td> </tr> <tr> <td>001:</td> <td>Wide</td> <td>FM Wide</td> </tr> <tr> <td>010:</td> <td>Normal</td> <td>QPSK System B/G/L/D/K</td> </tr> <tr> <td>011:</td> <td>Wide</td> <td>QPSK System B/G/L/D/K</td> </tr> <tr> <td>100:</td> <td>Normal</td> <td>FM Wide</td> </tr> <tr> <td>101:</td> <td>Wide</td> <td>FM Normal</td> </tr> <tr> <td>110:</td> <td>Normal</td> <td>QPSK System I</td> </tr> <tr> <td>111:</td> <td>Wide</td> <td>QPSK System I</td> </tr> </tbody> </table>		<u>CH1 FM</u>	<u>CH2 FM/QPSK</u>	000:	Normal	FM Normal	001:	Wide	FM Wide	010:	Normal	QPSK System B/G/L/D/K	011:	Wide	QPSK System B/G/L/D/K	100:	Normal	FM Wide	101:	Wide	FM Normal	110:	Normal	QPSK System I	111:	Wide	QPSK System I
	<u>CH1 FM</u>	<u>CH2 FM/QPSK</u>																											
000:	Normal	FM Normal																											
001:	Wide	FM Wide																											
010:	Normal	QPSK System B/G/L/D/K																											
011:	Wide	QPSK System B/G/L/D/K																											
100:	Normal	FM Wide																											
101:	Wide	FM Normal																											
110:	Normal	QPSK System I																											
111:	Wide	QPSK System I																											

**DEMOD\_STAT****Demodulator Detection Status Register**

Address: 0Dh

Type: R

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	QPSK_LK	FM2_CAR	FM2_SQ	FM1_CAR	FM1_SQ

Bit Name	Reset	Function
Bit [7:5]	000	Reserved.
QPSK_LK	0	<b>QPSK Lock Detection Flag</b> 0: Not detected 1: Detected
FM2_CAR	0	<b>Channel 2 FM/AM Carrier Detection Flag</b> 0: Not detected 1: Detected
FM2_SQ	0	<b>Channel 2 FM Squelch Detection Flag</b> 0: Not detected 1: Detected
FM1_CAR	0	<b>Channel 1 FM/AM Carrier Detection Flag</b> 0: Not detected 1: Detected
FM1_SQ	0	<b>Channel 1 FM Squelch Detection Flag</b> 0: Not detected 1: Detected

*Note: These registers allow direct access to the demodulator signal detectors.*

**AGC\_CTRL****IF AGC Control Register**

Address: 0Eh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
AGC_CMD	0	0	AGC_REF[2:0]		AGC_CST[1:0]		

Bit Name	Reset	Function																				
AGC_CMD	0	<p><b>Automatic Gain Control Command Mode</b></p> <p>Normally set to 0 enabling automatic mode. For L/L' standards, the AGC should be switched off due to the presence of the AM sound carrier. In this case, a fixed gain value should be set using the AGCS register.</p> <p>0: Automatic mode. AGC controlled by the Autostandard function. (Default) 1: Manual/Forced mode</p>																				
Bits[6:5]	00	Reserved.																				
AGC_REF[2:0]	100	<p>This bitfield is used to defines the clipping level which adjusts the allowable proportion of samples at the input of the ADC which will be clipped. The AGC tries to maximize the use of the full scale range of the ADC. The default setting gives a ratio of 1/256.</p> <table border="0"> <thead> <tr> <th colspan="2"><u>Clipping Ratio</u></th> <th colspan="2"><u>Clipping Ratio</u></th> </tr> </thead> <tbody> <tr> <td>000:</td> <td>1/16 (Single carrier)</td> <td>100:</td> <td>1/256 (Default)</td> </tr> <tr> <td>001:</td> <td>1/32</td> <td>101:</td> <td>1/512</td> </tr> <tr> <td>010:</td> <td>1/64</td> <td>110:</td> <td>1/1024</td> </tr> <tr> <td>011:</td> <td>1/128</td> <td>111:</td> <td>1/2048 (Multiple carriers)</td> </tr> </tbody> </table>	<u>Clipping Ratio</u>		<u>Clipping Ratio</u>		000:	1/16 (Single carrier)	100:	1/256 (Default)	001:	1/32	101:	1/512	010:	1/64	110:	1/1024	011:	1/128	111:	1/2048 (Multiple carriers)
<u>Clipping Ratio</u>		<u>Clipping Ratio</u>																				
000:	1/16 (Single carrier)	100:	1/256 (Default)																			
001:	1/32	101:	1/512																			
010:	1/64	110:	1/1024																			
011:	1/128	111:	1/2048 (Multiple carriers)																			
AGC_CST[1:0]	01	<p><b>AGC Time Constant</b></p> <p>This is the time constant between each step of 1.5 dB by the AGC.</p> <table border="0"> <thead> <tr> <th colspan="2"><u>Step Duration (ms)</u></th> </tr> </thead> <tbody> <tr> <td>00</td> <td>1.33</td> </tr> <tr> <td>01</td> <td>2.66</td> </tr> <tr> <td>10</td> <td>5.33</td> </tr> <tr> <td>11</td> <td>10.66</td> </tr> </tbody> </table>	<u>Step Duration (ms)</u>		00	1.33	01	2.66	10	5.33	11	10.66										
<u>Step Duration (ms)</u>																						
00	1.33																					
01	2.66																					
10	5.33																					
11	10.66																					

**AGC\_GAIN****IF AGC Control and Status Register**

Address: 0Fh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	AGC_ERR[4:0]				SIG_OVER	SIG_UNDER	

Bit Name	Reset	Function
Bit 7	0	Reserved.

Bit Name	Reset	Function
AGC_ERR[4:0]	00000	<b>Amplifier Gain Control</b> This is the Gain Control value of AGC. There are 20 steps of +1.5 dB (see Note below). 00000: Gain-min 10100: Gain-min + 30 db 11111: Gain-min + 30 db
SIG_OVER	0	<b>AGC Input Signal Upper Threshold</b> 0: Normal signal 1: Signal too large and AGC is overloaded
SIG_UNDER	0	<b>AGC Input Signal Lower Threshold</b> 0: Normal signal 1: Signal too small and AGC is underloaded  When the AGC is in Automatic mode (AGC_CMD = 0), bits SIG_OVER and SIG_UNDER indicate if the input signal is too small/large and the AGC is under/overloaded. This is useful when setting the STV82x7 SIF input level.

*Note: When **AGC\_CMD = 0**, **AGC\_ERR[4:0]** can be read -- indicating the input level. It can also be written to -- presetting the AGC level which will then adjust itself to the final value.*

*When **AGC\_CMD = 1**, the AGC is off and writing to **AGC\_ERR[4:0]** directly controls the AGC amplifier gain. Reading AGC\_ERR just confirms the fixed value.*

### DC\_ERR\_IF

### DC Offset Status for IF ADC

Address: 10h

Type: R

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

DC\_ERR[7:0]

Bit Name	Reset	Function
DC_ERR[7:0]	00000000	DC offset error of IF ADC output

## 12.5 Demodulator Channel 1

### CARFQ1H, CARFQ1M, CARFQ1L      Channel 1 Carrier DCO Frequency

Address: 12h to 14h

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

CARFQ1[23:16], CARFQ1[15:8], CARFQ1[7:0]

Bit Name	Reset	Function
CARFQ1[23:16]	00111110	Channel 1 DCO Carrier Frequency (8 MSBs)
CARFQ1[15:8]	10000000	Channel 1 DCO Carrier Frequency
CARFQ1[7:0]	00000000	Channel 1 DCO Carrier Frequency (8 LSBs), see <a href="#">Table 10</a> .

Table 10: Mono Carrier Frequencies by System

System	Mono Carrier Freq. (MHz)	CARFQ1[23:0] (dec)	CARFQ1[23:0]
M/N	4.5	3072000	2EE000h
B/G	5.5	3754667	394AABh
I	6.0	4096000	3E8000h
L	6.5	4453717	43F555h
D/K/K1/K2	6.5	4437333	43B555h

Note: Carrier Freq:  $CARFQ1(dec) \cdot f_S / 2^{24}$  with  $f_S = 24.576$  MHz (crystal oscillator frequency independent)

**FIR1C[0:7]****Channel 1 FIR Coefficients**

Address: 15h to 1Ch

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

FIR1C0[7:0] to FIR1C7[7:0]

Table 11: Channel 1 FIR Coefficients

Bitfield	Description							
	FM 27 kHz <sup>1</sup>	FM 50 kHz <sup>2</sup>	FM 75 kHz	FM 100 kHz	FM 200 kHz	FM 350 kHz	FM 500 kHz	AM
FIR1C0[7:0]	FFh	00h	01h	FFh	00h	02h	01h	00h
FIR1C1[7:0]	FEh	FEh	03h	00h	01h	01h	00h	FEh
FIR1C2[7:0]	FEh	FCh	02h	05h	01h	FCh	04h	FDh
FIR1C3[7:0]	00h	FDh	FCh	02h	FCh	03h	FAh	FEh
FIR1C4[7:0]	06h	02h	F8h	F8h	08h	04h	05h	04h
FIR1C5[7:0]	0Eh	0Dh	01h	F9h	F6h	F2h	00h	0Dh
FIR1C6[7:0]	16h	18h	18h	15h	F8h	06h	F2h	16h
FIR1C7[7:0]	1Bh	1Fh	2Dh	35h	4Ah	43h	4Dh	1Dh

1. Default mode for M/N standard.
2. Default mode for B/G/I/D/K standards

**ACOEFF1****Channel 1 Baseband PLL Loop Filter Proportional Coefficient**

Address: 1Dh

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

ACOEFF1[7:0]

Bit Name	Reset	Function
ACOEFF1[7:0]	00100011	Used to program the Proportional Coefficient of the baseband PLL loop filter (Channel 1) Defines the damping factor of the loop. For values, refer to <a href="#">Table 12</a> .

**BCOEFF1****Channel 1 Baseband PLL Loop Filter Integral Coefficient & DCO Gain**

Address: 1Eh

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

BCOEFF1[7:0]

Bit Name	Reset	Function
BCOEFF1[7:0]	00010010	Used to program the Integral Coefficient of the baseband PLL loop filter and DCO gain Defines the bandwidth of the loop. For values, refer to <a href="#">Table 12</a> .

**Table 12: Baseband PLL Loop Filter Adjustment (FM Mode)**

FM Mode	Small	Standard	Medium	Wide <sup>1</sup>	A2 Standard
ACOEFF	10h	22h	2Ch	2Ch	10h
BCOEFF	1Ah	12h	0Ah	0Ah	11h
FM_DEV max (kHz)	62.5	125	250	500	125
DCO Range (kHz)	96	192	384	768	192

1. Refer to DEMOD\_MODE[2:0] bits in the DEMOD\_CTRL register.

**Note: 1 FM Pre-scale has to be adjusted depending on the chosen FM Mode.****2 FM squelch threshold has to be adjusted depending on the chosen FM Mode.**

**CRF1****Channel 1 Baseband PLL Demodulator Offset**

Address: 1Fh

Type: R

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

CRF1[7:0]

Bit Name	Reset	Function
CRF1[7:0]	(00000000)	<b>Channel 1 Carrier Recovery Frequency</b> Displays the instantaneous frequency offset of the Channel 1 Baseband PLL Demodulator.

**CETH1****Channel 1 FM/AM Carrier Level Threshold**

Address: 20h

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

CETH1[7:0]

Bit Name	Reset	Function																				
CETH1[7:0]	00100000	This register is used to compare the carrier level in the channel and the threshold value. This level is measured after the channel filter and is relative to the full scale reference level (0 dB). This is used as part of the validation of an FM signal, if the carrier level is below the threshold, the signal is considered to be non-valid.  <table border="1"> <thead> <tr> <th><u>CETH</u></th> <th><u>Threshold (dB)</u></th> <th><u>CETH</u></th> <th><u>Threshold (dB)</u></th> </tr> </thead> <tbody> <tr> <td>FFh</td> <td>-6</td> <td>10h</td> <td>-32 (Recommended Value)</td> </tr> <tr> <td>80h</td> <td>-12</td> <td>08h</td> <td>-38</td> </tr> <tr> <td>40h</td> <td>-18</td> <td>00h</td> <td>OFF (all carrier levels are accepted)</td> </tr> <tr> <td>20h</td> <td>-24 (Default)</td> <td></td> <td></td> </tr> </tbody> </table>	<u>CETH</u>	<u>Threshold (dB)</u>	<u>CETH</u>	<u>Threshold (dB)</u>	FFh	-6	10h	-32 (Recommended Value)	80h	-12	08h	-38	40h	-18	00h	OFF (all carrier levels are accepted)	20h	-24 (Default)		
<u>CETH</u>	<u>Threshold (dB)</u>	<u>CETH</u>	<u>Threshold (dB)</u>																			
FFh	-6	10h	-32 (Recommended Value)																			
80h	-12	08h	-38																			
40h	-18	00h	OFF (all carrier levels are accepted)																			
20h	-24 (Default)																					

**SQTH1****Channel 1 FM Squelch Threshold Register**

Address: 21h

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

SQTH1[7:0]



Bit Name	Reset	Function												
SQTH1[7:0]	00111100	<p>The Squelch Detector measures the level of high frequency noise (&gt; 40 kHz) and compares it to the threshold level (SQTH). If the level is below this value, the S/N of the FM signal is considered to be acceptable. Values are given for FM with standard deviation.</p> <table border="1"> <thead> <tr> <th>SQTH</th> <th>S/N (dB)</th> </tr> </thead> <tbody> <tr> <td>FAh</td> <td>0</td> </tr> <tr> <td>77h</td> <td>10</td> </tr> <tr> <td>3Ch</td> <td>15 (Default)</td> </tr> <tr> <td>23h</td> <td>20</td> </tr> <tr> <td>19h</td> <td>25</td> </tr> </tbody> </table>	SQTH	S/N (dB)	FAh	0	77h	10	3Ch	15 (Default)	23h	20	19h	25
SQTH	S/N (dB)													
FAh	0													
77h	10													
3Ch	15 (Default)													
23h	20													
19h	25													

**Note:** FM squelch threshold has to be adjusted depending on the chosen FM Mode.

### CAROFFSET1

### Channel 1 DCO Carrier Offset Compensation

Address: 22h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
CAROFFSET1[7:0] (S)							

Bit Name	Reset	Function
CAROFFSET1[7:0]	00000000	<p>This value is used to correct the carrier frequency offset of the incoming IF signal. Automatic frequency control in FM mode can be implemented by registers <a href="#">DC_REMOVAL_L</a> and <a href="#">DC_REMOVAL_R</a>.</p> <p>A DCO frequency offset (in two's complement format) is added to the pre-programming value by AUTOTSD in the CARFQ1 registers (corresponding to the standard IF carrier frequency). The programmable carrier offset ranges from -192 kHz to +190.5 kHz with a resolution of 1.5 kHz.</p> <p>For standard FM deviation, the value displays by <a href="#">DC_REMOVAL_L</a> and <a href="#">DC_REMOVAL_R</a> can be directly loaded in CAROFFSET1 to exactly compensate the carrier offset on Channel 1.</p>

## 12.6 Demodulator Channel 2

### IAGCR

### Channel 2 Internal AGC Reference for QPSK

Address: 25h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
IAGC_REF[7:0]							

Bit Name	Reset	Function
IAGC_REF[7:0]	10001000	Sets the mean value of the internal AGC, used for QPSK demodulation. The default setting corresponds to half full scale amplitude at the baseband PLL input.

## IAGCC

## Channel 2 Internal AGC Time Constant for QPSK

Address: 26h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
IAGC_OFF	FAR_FLT_EN	MONO_FLT_EN	BG_SEL	MONO_PROG	IAGC_CST[2:0]		

Bit Name	Reset	Function																								
IAGC_OFF	0	AGC Disable 0: Internal AGC is active 1: Internal AGC is disabled																								
FAR_FLT_EN	0	1: Enable Farrow filter for NICAM																								
MONO_FLT_EN	0	1: Enable Mono filter for NICAM																								
BG_SEL	0	1: BG NICAM Mono filter selected																								
MONO_PROG	0	1: Enable programming of Mono filter																								
IAGC_CST[2:0]	011	<p><b>Internal AGC Programmable Step Constant.</b></p> <p>These bits control the time per step (values given for QPSK mode). The default value defines the optimum trade-off between fast settling time (for the fastest NICAM identification) and the noise immunity (minimum BER degradation)</p> <p style="text-align: center;"><u>Step time (us) Time Response (ms)</u></p> <table border="1" style="margin-left: auto; margin-right: auto;"> <tr><td>000</td><td>703</td><td>128</td></tr> <tr><td>001</td><td>352</td><td>64</td></tr> <tr><td>010</td><td>176</td><td>32</td></tr> <tr><td>011</td><td>88</td><td>16</td></tr> <tr><td>100</td><td>44</td><td>8</td></tr> <tr><td>101</td><td>22</td><td>4</td></tr> <tr><td>110</td><td>11</td><td>2</td></tr> <tr><td>111</td><td>5.5</td><td>0.82</td></tr> </table>	000	703	128	001	352	64	010	176	32	011	88	16	100	44	8	101	22	4	110	11	2	111	5.5	0.82
000	703	128																								
001	352	64																								
010	176	32																								
011	88	16																								
100	44	8																								
101	22	4																								
110	11	2																								
111	5.5	0.82																								

## IAGCS

## Channel 2 Internal AGC Status for QPSK

Address: 27h

Type: R

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
IAGC_CTRL[7:0]							

Bit Name	Reset	Function
IAGC_CTRL[7:0]	00000000	Indicates the value of the internal AGC gain control

**CARFQ2H, CARFQ2M, CARFQ2L Channel 2 Carrier DCO Frequency**

Address: 28H to 2Ah

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

CARFQ2[23:16], CARFQ2[15.8], CARFQ2[7:0]

Bit Name	Reset	Function
CARFQ2[23:16]	01000100	Channel 2 DCO Carrier Frequency (8 MSBs)
CARFQ2[15.8]	01000000	Channel 2 DCO Carrier Frequency
CARFQ2[7:0]	00000000	Channel 2 DCO Carrier Frequency (8 LSBs) See <a href="#">Table 13</a> .

**Table 13: Stereo Carrier Frequencies by System**

System	Stereo Carrier Freq. (MHz)	CARFQ2[23:0] (Dec)	CARFQ2[23:0]
M/N A2+	4.724212	3225062	3135E6h
B/G NICAM	5.85	3993600	3CF000h
BG A2	5.7421875	3920000	3BD080h
I NICAM	6.552	4472832	444000h
L NICAM	5.85	3993600	3CF000h
DK NICAM	5.85	3993600	3CF000h
DK1 A2*	6.258125	4272000	412F80h
DK2 A2*	6.7421875	4602667	463B2Bh
DK3 A2*	5.7421875	3920000	3BD080h

**FIR2C[0:7]****Channel 2 FIR Coefficients**

Address: 2Bh to 32h

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

FIR2C0[7:0] to FIR2C7[7:0]

**Table 14: Channel 2 FIR Coefficients**

Bitfield	Description			
	FM 27 kHz	FM 50 kHz	QPSK 40%	(reset state) QPSK100%
FIR2C0[7:0]	FFh	00h	00h	00h
FIR2C1[7:0]	FEh	FEh	00h	00h
FIR2C2[7:0]	FEh	FCh	FFh	00h

Table 14: Channel 2 FIR Coefficients

Bitfield	Description			
	FM 27 kHz	FM 50 kHz	QPSK 40%	(reset state) QPSK100%
FIR2C3[7:0]	00h	FDh	03h	00h
FIR2C4[7:0]	06h	02h	00h	FFh
FIR2C5[7:0]	0Eh	0Dh	F4h	04h
FIR2C6[7:0]	16h	18h	0Ah	14h
FIR2C7[7:0]	1Bh	1Fh	3Dh	25h

**ACOEFF2****Channel 2 Baseband PLL Loop Filter Proportional Coefficient**

Address: 33h

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

ACOEFF2[7:0]

Bit Name	Reset	Function
ACOEFF2[7:0]	10010000	This value defines the loop clamping factor used to program the Proportional Coefficient of the baseband PLL loop filter (Channel 2). See <a href="#">Table 15</a> and <a href="#">Table 16</a> .

**BCOEFF2****Channel 2 Baseband PLL Loop Filter Integral Coefficient & DCO Gain**

Address: 34h

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

BCOEFF2[7:0]

Bit Name	Reset	Function
BCOEFF2[7:0]	10101100	This value defines the loop bandwidth used to program the Integral Coefficient of the Baseband PLL loop filter and DCO gain. See <a href="#">Table 15</a> and <a href="#">Table 16</a> .

Table 15: Baseband PLL Loop Filter Adjustments (FM Mode)

FM mode	Small	Standard	Mid	Wide	A2 standard
ACOEFF	10h	22h	2Ch	2Ch	10h
BCOEFF	1Ah	12h	0Ah	0Ah	11h

**Table 15: Baseband PLL Loop Filter Adjustments (FM Mode)**

FM mode	Small	Standard	Mid	Wide	A2 standard
FM_DEV max (kHz)	62.5	125	250	500	125
DCO Range (kHz)	96	192	384	768	192

**Table 16: Baseband PLL Loop Filter Adjustments (QPSK Mode)**

QPSK mode	Small	Medium	Large	Extra-large
ACOEFF	90h	90h	90h	90h
BCOEFF	ACh	A3h	9Ah	91h
DCO_DEV max (kHz)	2.84375	5.6875	11.375	22.75

**SCOEFF**

**Channel 2 Symbol Tracking Loop Coefficients**

Address: 35h

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0



Bit Name	Reset	Function
SCOEFF[7:0]	00011100	This value is used to program the proportional and integral coefficients of the QPSK Symbol tracking loop. See <a href="#">Table 17</a> and <a href="#">Table 18</a> .

**Table 17: QPSK System - BG/L/DK Standards (40% Roll-off)**

	Extra-Small	Small	Medium	Large	Extra-Large	Open Loop
SCOEFF	1Eh	25h	24h	26h	2Ah	80h

**Table 18: QPSK System - I Standard (100% Roll-off)**

	Extra-Small	Small	Medium	Large	Extra-Large
SCOEFF	16h	1Dh	1Ch	23h	22h

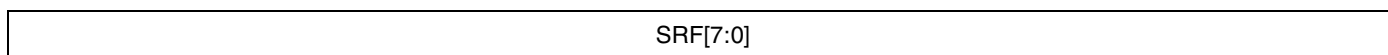
**SRF**

**Channel 2 Symbol Tracking Loop Frequency**

Address: 36h

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0



Bit Name	Reset	Function
SRF[7:0]	00000000	Displays in two's complement format the frequency deviation between the incoming NICAM bitstream and the quartz clocks. The maximum error is $\pm 250$ ppm.

**CRF2****Channel 2 Baseband PLL Demodulator Offset**

Address: 37h

Type: R

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
CRF2[7:0]							

Bit Name	Reset	Function
CRF2[7:0]	00000000	<b>Channel 2 Carrier Recovery Frequency.</b> Displays the instantaneous frequency offset of the Channel 2 Baseband PLL

**CAROFFSET2****Channel 2 DCO Carrier Offset Compensation**

Address: 3Ah

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
CAROFFSET2[7:0] (S)							

Bit Name	Reset	Function
CAROFFSET2[7:0]	00000000	This value is used to correct the carrier frequency offset of the incoming IF signal. Automatic frequency control in FM mode can be implemented by registers <a href="#">DC_REMOVAL_L</a> and <a href="#">DC_REMOVAL_R</a> .  A DCO frequency offset (in two's complement format) is added to the pre-programming value by AUTOTSD in the CARFQ2 registers (corresponding to the standard IF carrier frequency). The programmable carrier offset ranges from -192 kHz to +190.5 kHz with a resolution of 1.5 kHz.  For standard FM deviation, the value displayed by register <a href="#">DC_REMOVAL_R</a> can be directly loaded in register <a href="#">CAROFFSET2</a> to exactly compensate the carrier offset on Channel 2.

**12.7 NICAM Registers****NICAM\_CTRL****NICAM Decoder Control Register**

Address: 3Dh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	DIF_POL	ECT	MAE

Bit Name	Reset	Function
Bits[7:3]	00000	Reserved.
DIF_POL	0	0: No polarity inversion (Default) 1: Polarity inversion of the differential decoding
ECT	0	<b>Error Counter Timer:</b> Defines the NICAM error measurement period 0: 128 ms (Default) 1: 64 ms
MAE	0	<b>Max. Allowed Errors.</b> Defines the NICAM error decoding for mute function. 0: 511 Max (Default) 1: 255 Max

**NICAM\_BER****NICAM Bit Error Rate Register**

Address: 3Eh

Type: R

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
ERROR[7:0]							

Bit Name	Reset	Function
ERROR[7:0]	00000000	<b>NICAM Error Counter Value</b>

**NICAM\_STAT****NICAM Detection Status Register**

Address: 3Fh

Type: R

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
NIC_DET	F_MUTE	LOA	CBI[3:0]			NIC_MUTE	

Bit Name	Reset	Function
NIC_DET	0	<b>NICAM Signal Detect</b> 0: NICAM signal no detected 1: NICAM signal detected
F_MUTE	0	<b>Frame Mute</b> 0: No mute 1: Mute due to Superframe Alignment Loss
LOA	0	<b>Loss of Frame Alignment Word (FAW)</b> 0: No Alignment Lost 1: Frame Alignment Word Lost
CBI[3:0]	0000	Indicates the received NICAM control bits
NIC_MUTE	0	Indicates the NICAM decoder mute

## 12.8 Stereo Mode

## ZWT\_CTRL

## Zweiton Detector Control Register

Address: 40h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
LRST_TONE_OFF	STD_MODE	THRESH[3:0]			TSCTRL[1:0]		

Bit Name	Reset	Function																																				
LRST_TONE_OFF	0	<b>Control of the reset of the tone detector</b> 0: Periodical reset of tone detection enabled 1: Periodical reset of tone detection disabled																																				
STD_MODE_C	0	0: German standard (Default) 1: Korean standard																																				
THRESH[3:0]	1100	Defines the threshold of the detector for pilot and tone frequencies.  <table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th colspan="2" style="text-align: center;"><u>Level</u> (% of the mid scale)</th> <th colspan="2" style="text-align: center;"><u>Level</u> (% of the mid scale)</th> </tr> </thead> <tbody> <tr><td>0000</td><td style="text-align: center;">0</td><td>1000</td><td style="text-align: center;">50</td></tr> <tr><td>0001</td><td style="text-align: center;">6.25</td><td>1001</td><td style="text-align: center;">56.25</td></tr> <tr><td>0010</td><td style="text-align: center;">12.5</td><td>1010</td><td style="text-align: center;">62.5</td></tr> <tr><td>0011</td><td style="text-align: center;">18.75</td><td>1011</td><td style="text-align: center;">68.75</td></tr> <tr><td>0100</td><td style="text-align: center;">25</td><td>1100 (Default)</td><td style="text-align: center;">75</td></tr> <tr><td>0101</td><td style="text-align: center;">31.25</td><td>1101</td><td style="text-align: center;">81.25</td></tr> <tr><td>0110</td><td style="text-align: center;">37.5</td><td>1110</td><td style="text-align: center;">87.5</td></tr> <tr><td>0111</td><td style="text-align: center;">43.75</td><td>1111</td><td style="text-align: center;">93.75</td></tr> </tbody> </table>	<u>Level</u> (% of the mid scale)		<u>Level</u> (% of the mid scale)		0000	0	1000	50	0001	6.25	1001	56.25	0010	12.5	1010	62.5	0011	18.75	1011	68.75	0100	25	1100 (Default)	75	0101	31.25	1101	81.25	0110	37.5	1110	87.5	0111	43.75	1111	93.75
<u>Level</u> (% of the mid scale)		<u>Level</u> (% of the mid scale)																																				
0000	0	1000	50																																			
0001	6.25	1001	56.25																																			
0010	12.5	1010	62.5																																			
0011	18.75	1011	68.75																																			
0100	25	1100 (Default)	75																																			
0101	31.25	1101	81.25																																			
0110	37.5	1110	87.5																																			
0111	43.75	1111	93.75																																			
TSCTRL[1:0]	00	Defines both the detection time and the error probability (reliability of the detection).  <table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: center;"><u>Sample Accumulation</u></th> <th style="text-align: center;"><u>Decision Count</u></th> <th style="text-align: center;"><u>Time (ms)</u></th> <th style="text-align: center;"><u>Error Probability</u></th> </tr> </thead> <tbody> <tr><td>00</td><td style="text-align: center;">1024</td><td style="text-align: center;">2</td><td style="text-align: center;">256</td><td style="text-align: center;"><math>10^{-4}</math></td></tr> <tr><td>01 (Default)</td><td style="text-align: center;">1024</td><td style="text-align: center;">3</td><td style="text-align: center;">384</td><td style="text-align: center;"><math>10^{-6}</math></td></tr> <tr><td>10</td><td style="text-align: center;">2048</td><td style="text-align: center;">2</td><td style="text-align: center;">512</td><td style="text-align: center;"><math>10^{-7}</math></td></tr> <tr><td>11</td><td style="text-align: center;">2048</td><td style="text-align: center;">3</td><td style="text-align: center;">768</td><td style="text-align: center;"><math>10^{-9}</math></td></tr> </tbody> </table>	<u>Sample Accumulation</u>	<u>Decision Count</u>	<u>Time (ms)</u>	<u>Error Probability</u>	00	1024	2	256	$10^{-4}$	01 (Default)	1024	3	384	$10^{-6}$	10	2048	2	512	$10^{-7}$	11	2048	3	768	$10^{-9}$												
<u>Sample Accumulation</u>	<u>Decision Count</u>	<u>Time (ms)</u>	<u>Error Probability</u>																																			
00	1024	2	256	$10^{-4}$																																		
01 (Default)	1024	3	384	$10^{-6}$																																		
10	2048	2	512	$10^{-7}$																																		
11	2048	3	768	$10^{-9}$																																		

## ZWT\_TIME

## Zweiton Detector Timing Register

Address: 41h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	ZWT_TIME[2:0]		

Bit Name	Reset	Function
Bit [7:3]	00000	Reserved.



Bit Name	Reset	Function
ZWT_TIME[2:0]	100	Defines the period (duration) of the reset tone used for tone detection system reset. 000: 256 ms                      100: 1280 ms 001: 512 ms                      101: 1536 ms 010: 768 ms                      110: 1792 ms 011: 1024 ms                      111: 2040 ms

**ZWT\_STAT****Zweiton Status Register**

Address: 42h

Type: R

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
LRST_TONE_OFF	0	0	0	ZW_STAT_RDY	ZW_DET	ZW_ST	ZW_DM

Bit Name	Reset	Function
LRST_TONE_OFF	0	<b>Indicates the status of the control bit programmed in the reg ZWT-CTRL</b> 0: Periodical reset of tone detection enabled 1: Periodical reset of tone detection disabled
Bits[6:4]	000	Reserved.
ZW_STAT_RDY	0	Periodic flag indicating when the tone detection flags are updated and ready to be read
ZW_DET	0	Pilot Detection Flag
ZW_ST	0	Stereo Tone Detection Flag
ZW_DM	0	Dual Mono Tone Detection Flag

**12.9 Analog Control****ADC\_CTRL**

Address: 56h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
I2S_DATA0_CTRL[1:0]	0	0	ADC_POWER_UP	ADC_INPUT_SEL[2:0]			

Bit Name	Reset	Function
I2S_DATA0_CTRL[1:0]	00	00 = SCART 01 = L, R 10 = HP or Srnd 11 = C/Sub
Bits[7:4]	0000	Reserved.

Bit Name	Reset	Function
ADC_POWER_UP	1	<b>Control of the power up of the Audio ADC</b> 0: ADC in power down mode 1: Wake up of the ADC
ADC_INPUT_SEL [2:0]	000	<b>Selection of the ADC input signal</b> 000: SCART 1 (Default)                      011: SCART 4 001: SCART 2                                      100: Mono input 010: SCART 3                                      Other: reserved

### SCART1\_2\_OUTPUT\_CTRL

Address: 57h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
SC2_MUTE	SC2_OUTPUT_SEL[2:0]			SC1_MUTE	SC1_OUTPUT_SEL[2:0]		

Bit Name	Reset	Function
SC2_MUTE	1	<b>Mute command for the output SCART 2</b> 0: output not muted 1: output muted
SC2_OUTPUT_SEL[2:0]	010	<b>Selection of the output SCART 2 configuration:</b> 000: DSP    100: Input SCART 3 001: Mono input                                      101: Input SCART 4 010: Input SCART 1 (Default)                      Other: Reserved 011: Input SCART 2
SC1_MUTE	1	<b>Mute command for the output scart 1</b> 0: output not muted 1: output muted
SC1_OUTPUT_SEL[2:0]	000	<b>Selection of the output SCART 1 configuration:</b> 000: DSP (Default)                                      100: Input SCART 3 001: Mono input    101: Input SCART 4 010: Input SCART 1                                      Other: Reserved 011: Input SCART 2

### SCART3\_OUTPUT\_CTRL

Address: 58h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	SC3_MUTE	SC3_OUTPUT_SEL[2:0]		

Bit Name	Reset	Function
Bits[7:4]	0000	Reserved.
SC3_MUTE	1	<b>Mute command for the output SCART 3</b> 0: output not muted 1: output muted
SC3_OUTPUT_SEL[2:0]	011	<b>Selection of the output SCART 3 configuration:</b> 000: DSP 001: Mono input 010: Input SCART 1 011: Input SCART 2 (Default) 100: Input SCART 3 101: Input SCART 4 Other: Reserved

## 12.10 Clocking 2

### FS2\_DIV

### FS2 I/O Divider Programming Register

Address: 5Ah

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	NDIV2[1:0]			SDIV2[2:0]		

Bit Name	Reset	Function
Bit [7:6]	0	Reserved.
NDIV2[1:0]	01	FS2 Input clock divider selection
Bit 4	0	Reserved.
SDIV2[2:0]	001	FS2 Output clock divider selection

### FS2\_MD

### FS2 Coarse Selection Register

Address: 5Bh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	MD2[4:0]				

Bit Name	Reset	Function
Bits[7:5]	000	Reserved.
MD2[4:0]	10000	FS2 Coarse Selection

**FS2\_PE\_H****FS2 Fine Selection Register (MSBs)**

Address: 5Ch

Type: R/W

Bit 7            Bit 6            Bit 5            Bit 4            Bit 3            Bit 2            Bit 1            Bit 0

PE\_H2[7:0]

Bit Name	Reset	Function
PE_H2[7:0]	0101 1100	FS2 Fine Selection (MSBs)

**FS2\_PE\_L****FS2 Fine Selection Register (LSBs)**

Address: 5Dh

Type: R/W

Bit 7            Bit 6            Bit 5            Bit 4            Bit 3            Bit 2            Bit 1            Bit 0

PE\_L2[7:0]

Bit Name	Reset	Function
PE_L2[7:0]	0010 1001	FS2 Fine Selection (LSBs)

**12.11 DSP Control****HOST\_CMD****DSP Hardware Control Register**

Address: 80h

Type: R/W

Bit 7            Bit 6            Bit 5            Bit 4            Bit 3            Bit 2            Bit 1            Bit 0

IT_IN_DSP	0	0	0	0	HW_RESET		
-----------	---	---	---	---	----------	--	--

Bit Name	Reset	Function
IT_IN_DSP	0	Valid I2C table.
Bits[6:3]	0000	Reserved.
HW_RESET	0	DSP Hardware run when set, see <a href="#">Figure 31</a>
Bits[1:0]	00	Reserved.

**IRQ\_STATUS****IRQ Status Register**

Address: 81h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
IRQ7	IRQ6	IRQ5	IRQ4	IRQ3	IRQ2	IRQ1	IRQ0

Bit Name	Reset	Function
Bits[7:4]	0000	Reserved.
IRQ3	0	Unmute HP/Srmd DAC IRQ
IRQ2	0	HP connection/deconnection IRQ
IRQ1	0	I2S lock lost IRQ
IRQ0	0	Auto-Standard IRQ

**SOFT\_VERSION****Embedded Software Version Register**

Address: 82h

Type: R

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
SOFT_VERSION[7:0]							

Bit Name	Reset	Function
SOFT_VERSION[7:0]	0000 0002	Version of the Embedded software.

**ONCHIP\_ALGOS**

Address: 83h

Type: R

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	PRO_LOGIC_SELECT	NICAM	I2S_INPUT	TRUBASS	TRU SURROUND	PRO_LOGIC	MULTICHANNEL

Bit Name	Reset	Function
Bit 7	0	Reserved.
PRO_LOGIC_SELECT	0	0: Dolby Pro Logic I 1: Dolby Pro Logic II
NICAM	0	NICAM Demodulator is present when set.

Bit Name	Reset	Function
I2S_INPUT	0	0: 1 I2S input 1: 3 I2S inputs
DIALOG_CLARITY	0	SRS Dialog Clarity algorithm is present when set.
TRUBASS	0	SRS Trubass algorithm is present when set.
TRUSURROUND	0	SRS Trusurround algorithm is present when set.
PRO_LOGIC	0	Dolby Pro Logic algorithm is present when set.
MULTICHANNEL	0	Multichannels output is present when set.

**DSP\_STATUS****DSP Status Register**

Address: 84h

Type: R

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	0	0	INIT_MEM

Bit Name	Reset	Function
Bits[7:1]	0000000	Reserved.
INIT_MEM	0	<b>DSP Initialization</b> 0: DSP is not initialized. 1: DSP is initialized.

**DSP\_RUN**

Address: 85h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	0	HOST_NO_INIT	HOST_RUN

Bit Name	Reset	Function
Bits[7:6]	00	Reserved
Bits[5:4]	00	Reserved
Bits[3:2]	00	Reserved
HOST_NO_INIT	0	0: I2C register table is initialized when we soft reset 1: I2C register table is not initialized when we soft reset
HOST_RUN	0	0: soft reset DSP 1: start DSP processing

## I2S\_IN\_CONFIG

I<sup>2</sup>S Configuration Register

Address: 86h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
LOCK_MODE_EN	0	SYNC	LRCLK_START	LRCLK_POLARITY	SCLK_POLARITY	DATA_CFG	I2S_MODE

Bit Name	Reset	Function
LOCK_MODE_EN	1	0: Disable Lock Mode for external I2S input 1: Enable Lock Mode for external I2S input
Bit 6	0	Reserved.
SYNC	0	I2S synchronisation: 0: Capture directly 1: Wait for synchro
LRCLK_START	0	according to LRCLK POLARITY, first data take: 0: Left 1: Right
LRCLK_POLARITY	0	Polarity of the left data
SCLK_POLARITY	1	0: Falling edge 1: Rising edge
DATA_CFG	1	0: LSB First 1: MSB First
I2S_MODE	0	0: Non standard mode 1: Standard mode (Refer to <a href="#">Figure 29</a> )

## AV\_DELAY

## Audio/Video Delay Register

Address: 89h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
DELAY_TIME							DELAY_ON

Bit Name	Reset	Function
DELAY_TIME	0000000	<b>Audio Delay Time (see <a href="#">Table 19</a>)</b> 0000000: 0 ms ... 0111100: 60 ms (48 kHz) ... 1011010: 90 ms (32 kHz)
DELAY_ON	0	Audio/video delay is enabled when set.

Note: AV\_DELAY acts on both LS and HP paths simultaneously (same delay)

**Table 19: Audio/Video Delay (LipSync) Configuration**

Input source	Register Value							Output									
	AV_DELAY (89h)		PCM_SRND_DELAY (A8h)		PCM_CENTER_DELAY (ACh)			LS_L		LS_R		HP_L/R		Scart_L		Scart_R	
	DELAY_TIME[6:0]	DELAY_ON	SNRD_DELAY[4:0]		CENTER_DELAY[3:0]			Source SIF	Source Scart	Source SIF	Source Scart	Source SIF	Source Scart	Source SIF	Source Scart	Source SIF	Source Scart
SIF or Scart (32Khz)	10110100	90	1	xxx00000	0	xxxx0000	0	90	90	90	90	90	90	0	0	0	0
	10110100	90	1	xxx00000	0	xxxx1010	10	60	60	60	60	60	60	0	0	0	0
	10110100	90	1	xxx11110	30	xxxx0000	0	60	60	60	60	60	60	0	0	0	0
	10110100	90	1	xxx11110	30	xxxx1010	10	60	60	60	60	60	60	0	0	0	0
Scart only (48Khz)	01111000	60	1	xxx00000	0	xxxx0000	0	-	60	-	60	-	60	-	0	-	0
	01111000	60	1	xxx00000	0	xxxx1010	10	-	30	-	30	-	30	-	0	-	0
	01111000	60	1	xxx11110	30	xxxx0000	0	-	30	-	30	-	30	-	0	-	0
	01111000	60	1	xxx11110	30	xxxx1010	10	-	30	-	30	-	30	-	0	-	0

Note: All audio delay values are in milliseconds.

## 12.12 Automatic Standard Recognition

### AUTOSTD\_CTRL

### Automatic Standard Recognition Control Register

Address: 8Ah

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	FORCE_SQUELCH	SINGLE_SHOT	DK_DEV[1:0]		LDK_SW

Bit Name	Reset	Function
Bits[7:5]	000	Reserved.
FORCE_SQUELCH	0	Allow to force squelch detection 0: FM squelch is taken into consideration for MONO detection 1: FM squelch is not taken into consideration for MONO detection
SINGLE_SHOT	0	<b>Single Shot Mode Selection</b> 0: Single Shot mode is not selected 1: Single Shot mode is selected <sup>1</sup>
DK_DEV[1:0]	00	Selects FM deviation configuration to take into account of overmodulation in DK_NICAM standard. 00: FM 50 kHz (Default)      10: FM 350 kHz 01: FM 200 kHz                11: FM 500 kHz
LDK_SW	1	Makes exclusive the auto search of DK/K1/K2/K3 and L/L' standard 0: DK/K1/K2/K3 standard auto-search / L/L' disabled 1: L/L' standard auto-search DK/K1/K2/K3 disabled



1. **Single Shot** mode can be used before disabling the Automatic Standard Recognition (Autostandard) to pre-program demodulator registers in a defined standard and reduce I<sup>2</sup>C programming in Manual mode

*Note:* Only standard deviation FM 50K kHz is compatible with other D/K1/K2/K3 standards in Automatic Standard Recognition Search mode.

FM deviation superior to 350 kHz will degrade strongly NICAM reception due to overlapping of FM and QPSK IF spectrum in DK-NICAM standard.

L/L' and DK/K1/K2/K3 standard cannot be discriminated in Automatic Standard Recognition Search mode because the same frequency is used for the mono IF carrier.

### AUTOSTD\_STANDARD\_DETECT Auto Standard Check Standard Register

Address: 8Bh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	NICAM_C4_OFF	NICAM_GAP_MODE	NICAM_MONO_IN	LDK_SCK	I_SCK	BG_SCK	MN_SCK

Bit Name	Reset	Function
NICAM_C4_OFF	0	0: Autostandard will consider the C4 bit for MONO backup 1: Autostandard will ignore the C4 bit for MONO backup
NICAM_GAP_MODE	1	0: NICAM, fast search 1: NICAM, slow search (no perturbations on LEFT channel in search mode)
NICAM_MONO_IN	0	0: the MONO backup for NICAM comes from internal demodulator 1: the MONO backup for NICAM comes from MONO input
LDK_SCK	1	<b>L/L' or D/K Mono Standard Enable</b> 0: Disabled 1: Enabled
I_SCK	1	<b>I Mono Standard Enable</b> 0: Disabled 1: Enabled
BG_SCK	1	<b>B/G Mono Standard Enable</b> 0: Disabled 1: Enabled
MN_SCK	1	<b>M/N Mono Standard Enable</b> 0: Disabled 1: Enabled

*Note:* Autostandard is off when all mono standards are disabled (LDK\_SCK = 0, I\_SCK = 0, BG\_SCK = 0 and MN\_SCK = 0).

**AUTOSTD\_STEREO\_DETECT Auto Standard Check Stereo Register**

Address: 8Ch

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
LDK_ZWT3	LDK_ZWT2	LDK_ZWT1	LDK_NIC	I_NIC	BG_ZWT	BG_NIC	MN_ZWT

Bit Name	Reset	Function
LDK_ZWT3	0	<b>D/K3 Zweiton (A2*) Stereo Standard Enable</b> 0: Disabled 1: Enabled
LDK_ZWT2	0	<b>D/K2 Zweiton (A2*) Stereo Standard Enable</b> 0: Disabled 1: Enabled
LDK_ZWT1	0	<b>D/K1 Zweiton (A2*) Stereo Standard Enable</b> 0: Disabled 1: Enabled
LDK_NIC	1	<b>D/K NICAM Stereo Standard Enable</b> 0: Disabled 1: Enabled
I_NIC	1	<b>I NICAM Stereo Standard Enable</b> 0: Disabled 1: Enabled
BG_ZWT	1	<b>B/G Zweiton (A2) Standard Enable</b> 0: Disabled 1: Enabled
BG_NIC	1	<b>B/G NICAM Standard Enable</b> 0: Disabled 1: Enabled
MN_ZWT	1	<b>M/N Zweiton (A2+) Standard Enable</b> 0: Disabled 1: Enabled

*Note:* Stereo standard covers all transmission modes (stereo or multi-language) of the NICAM or Zweiton (A2, A2\* or A2+) system.

**AUTOSTD\_TIMERS****Detection Time Out Register**

Address: 8Dh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
FM_TIME[1:0]		NICAM_TIME[2:0]			ZWEITON_TIME[2:0]		

Bit Name	Reset	Function
FM_TIME[1:0]	10	<b>FM/AM Detection Time-out</b> 00 : 16 ms            10: 48 ms (Default) 01: 32 ms            11: 64 ms
NICAM_TIME[2:0]	100	<b>NICAM Detection Time-out</b> 000: 96 ms            100: 224 ms (Default) 001: 128 ms           101: 256 ms 010: 160 ms           110: 288 ms 011: 192 ms           111: 320 ms
ZWEITON_TIME[2:0]	100	<b>Zweiton Detection Time-out</b> 000: forbids           100: 1280 ms (Default) 001: 512 ms           101: 1536 ms 010: 768 ms           110: 1792 ms 011: 1024 ms          111: 2040 ms

Note: The time-out default value is optimum and does not normally need to be changed.

**AUTOSTD\_STATUS****Detection Standard Status Register**

Address: 8Eh

Type: R

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
STEREO_ID	STEREO_OK	MONO_OK	AUTOSTD_ON	STEREO_SID[1:0]		MONO_SID[1:0]	

Bit Name	Reset	Function
STEREO_ID	0	Stereo Mode Detection flag activated when all of the following conditions are true: 1. Stereo standard coming from the demodulator is selected on the Loudspeakers output 2. Stereo transmission modes are: - Zweiton Stereo Carrier AND Stereo Modulation (indifferently German or Korean standard) - NICAM stereo with backup (CBI = 1000) - NICAM stereo with no backup (CBI = 0000) 3. Stereo is selected for loudspeaker output (bit LS_LANGUAGE[1:0])
AUTOSTD_ON	0	<b>Automatic Standard Recognition System Status</b> 0: Automatic Standard Recognition System is OFF 1: Automatic Standard Recognition System is ON
STEREO_SID[1:0]	00	Identification of the detected TV sound standard. See <a href="#">Table 20</a> .
MONO_SID[1:0]	00	
STEREO_OK	0	STEREO STANDARD DETECTED
MONO_OK	0	MONO STANDARD DETECTED

Table 20: TV Sound Standards

System	Mono Sound (MHz)	MONO_SID [1:0]	LDK_SW	DK_DEV [1:0]	Stereo Sound (MHz)	STEREO_SID [1:0]
M/N	4.5 (FM 27k)	00	X	XX	4.724 (Zweiton A2+)	00
B/G	5.5 (FM 50k)	01	X	XX	5.85 (NICAM 40%)	00
			X	XX	5.742 (Zweiton A2)	01
I	6.0 (FM 50k)	10	X	XX	6.552 (NICAM 100%)	00
L	6.5 (AM)	11	1	XX	5.85 (NICAM 40%)	00
D/K	6.5 (FM 50k)		0	00	5.85 (NICAM 40%)	00
	6.5 (FM 200k)			01		
	6.5 (FM 350k)			10		
	6.5 (FM 500k)			11		
D/K1/K2/K3	6.5 (FM 50k)	0	XX	5.85 (NICAM 40%)	00	
		0	XX	6.258 (Zweiton A2*)	01	
		0	XX	6.742 (Zweiton A2*)	10	
		0	XX	5.742 (Zweiton A2*)	11	

Note: X means don't care.

## 12.13 Audio Preprocessing and Selection Registers

### DC\_REMOVAL\_INPUT

### DC Removal Register

Address: 90h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	DC_SCART	DC_NICAM	DC_DEMOD

Bit Name	Reset	Function
Bits[7:3]	00000	Reserved.
DC_SCART	1	0: SCART input, DC removal inactive 1: SCART input, DC removal active
DC_NICAM	1	0: NICAM input, DC removal inactive 1: NICAM input, DC removal active
DC_DEMOD	1	0: FM input, DC removal inactive 1: FM input, DC removal active

**DC\_REMOVAL\_L****FM DC Offset Left Register**

Address: 91h

Type: R

Bit 7            Bit 6            Bit 5            Bit 4            Bit 3            Bit 2            Bit 1            Bit 0

DC\_REMOVAL\_L[7:0]

Bit Name	Reset	Function
DC_REMOVAL_L[7:0]	0000 0000	Displays (in two's complement format) the FM (or AM) DC offset level after demodulation on channel 1 (and removed automatically). In FM mode, the DC offset value gives a direct value of the carrier frequency offset which is used to compensate the DCO with the CAROFFSET1 value in the event of an out-of-standard offset. The range and the resolution depend upon the FM bandwidth programmed defined in register BCOEFF1. See <a href="#">Table 21</a> .

**DC\_REMOVAL\_R****FM DC Offset Right Register**

Address: 92h

Type: R

Bit 7            Bit 6            Bit 5            Bit 4            Bit 3            Bit 2            Bit 1            Bit 0

DC\_REMOVAL\_R[7:0]

Bit Name	Reset	Function
DC_REMOVAL_R[7:0]	0000 0000	Displays (in two's complement format) the FM (or AM) DC offset level after demodulation on channel 2 (and removed automatically). In FM mode, the DC offset value gives a direct value of the carrier frequency offset which is used to compensate the DCO with the CAROFFSET2 value in the event of an out-of-standard offset. The range and the resolution depend upon the FM bandwidth programmed defined in register BCOEFF2. See <a href="#">Table 21</a> .

**Table 21: DC\_REMOVAL\_L/R Range and Resolution**

FM mode	Range (kHz)	Resolution (kHz)
Small	± 96	0.750
Standard & A2 Standard	± 192	1.5
Medium	± 384	3
Large	± 768	6

**PRESCALE\_SELECT****AM/FM Prescaling Select Register**

Address: 93h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	0	0	AM_FM_SELECT

Bit Name	Reset	Function
Bits[7:1]	0000000	Reserved.
AM_FM_SELECT	0	0: FM prescale is applied to demodulator channels 1: AM prescale is applied to demodulator channels

**PRESCALE\_AM****AM Prescaling Register**

Address: 94h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	PRESCALE_AM						

Bit Name	Reset	Function																												
Bit 7	0	Reserved.																												
PRESCALE_AM[6:0]	0000000	-12 to + 24 dB AM prescaling to normalize the AM demodulated signal level before audio processing. Auto level control can be implemented by I2C software using the Peak Level Detector. (Default value = 0 dB)																												
		<table border="0"> <thead> <tr> <th></th> <th><u>G (dB)</u></th> <th></th> <th><u>G (dB)</u></th> </tr> </thead> <tbody> <tr> <td>0110000</td> <td>+24</td> <td>1101100</td> <td>-10</td> </tr> <tr> <td>0101111</td> <td>+23.5</td> <td>1101011</td> <td>-10.5</td> </tr> <tr> <td>0101110</td> <td>+23</td> <td>1101010</td> <td>-11</td> </tr> <tr> <td>0101101</td> <td>+22.5</td> <td>1101001</td> <td>-11.5</td> </tr> <tr> <td>0101100</td> <td>+22</td> <td>1101000</td> <td>-12</td> </tr> <tr> <td></td> <td>etc.</td> <td></td> <td></td> </tr> </tbody> </table>		<u>G (dB)</u>		<u>G (dB)</u>	0110000	+24	1101100	-10	0101111	+23.5	1101011	-10.5	0101110	+23	1101010	-11	0101101	+22.5	1101001	-11.5	0101100	+22	1101000	-12		etc.		
	<u>G (dB)</u>		<u>G (dB)</u>																											
0110000	+24	1101100	-10																											
0101111	+23.5	1101011	-10.5																											
0101110	+23	1101010	-11																											
0101101	+22.5	1101001	-11.5																											
0101100	+22	1101000	-12																											
	etc.																													

**PRESCALE\_FM****FM Prescaling Register**

Address: 95h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	PRESCALE_FM						

Bit Name	Reset	Function																												
Bit 7	0	Reserved.																												
PRESCALE_FM[6:0]	0001100	-12 to + 24 dB FM prescaling to normalize the FM demodulated signal level before audio processing. Auto level control can be implemented by I2C software using the Peak Level Detector. (Default value = +6 dB)																												
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	<u>G (dB)</u>		<u>G (dB)</u>																											
0110000	+24	1101100	-10																											
0101111	+23.5	1101011	-10.5																											
0101110	+23	1101010	-11																											
0101101	+22.5	1101001	-11.5																											
0101100	+22	1101000	-12																											
	etc.																													

**PRESCALE\_NICAM****NICAM Prescaling Register**

Address: 96h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	PRESCALE_NICAM						

Bit Name	Reset	Function																												
Bit 7	0	Reserved.																												
PRESCALE_NICAM[6:0]	011010	-6 to + 24 dB NICAM prescaling to normalize the NICAM demodulated signal level before audio processing. Auto level control can be implemented by I2C software using the Peak Level Detector. (Default value = +13 dB)																												
		<table> <thead> <tr> <th></th> <th><u>G (dB)</u></th> <th></th> <th><u>G (dB)</u></th> </tr> </thead> <tbody> <tr> <td>0110000</td> <td>+24</td> <td>1111000</td> <td>-4</td> </tr> <tr> <td>0101111</td> <td>+23.5</td> <td>1110111</td> <td>-4.5</td> </tr> <tr> <td>0101110</td> <td>+23</td> <td>1110110</td> <td>-5</td> </tr> <tr> <td>0101101</td> <td>+22.5</td> <td>1110101</td> <td>-5.5</td> </tr> <tr> <td>0101100</td> <td>+22</td> <td>1110100</td> <td>-6</td> </tr> <tr> <td></td> <td>etc.</td> <td></td> <td></td> </tr> </tbody> </table>		<u>G (dB)</u>		<u>G (dB)</u>	0110000	+24	1111000	-4	0101111	+23.5	1110111	-4.5	0101110	+23	1110110	-5	0101101	+22.5	1110101	-5.5	0101100	+22	1110100	-6		etc.		
	<u>G (dB)</u>		<u>G (dB)</u>																											
0110000	+24	1111000	-4																											
0101111	+23.5	1110111	-4.5																											
0101110	+23	1110110	-5																											
0101101	+22.5	1110101	-5.5																											
0101100	+22	1110100	-6																											
	etc.																													

**PRESCALE\_SCART****SCART Prescaling Register**

Address: 97h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	PRESCALE_SCART					

Bit Name	Reset	Function
Bit [7:6]	00	Reserved.

Bit Name	Reset	Function																												
PRESCALE_SCART[5:0]	0000000	-12 to + 12 dB SCART prescaling to normalize the SCART signal level before audio processing. Auto level control can be implemented by I2C software using the Peak Level Detector. (Default value = 0 dB)																												
		<table border="0"> <thead> <tr> <th></th> <th>G (dB)</th> <th></th> <th>G (dB)</th> </tr> </thead> <tbody> <tr> <td>011000</td> <td>+12</td> <td>101100</td> <td>-10</td> </tr> <tr> <td>010111</td> <td>+11.5</td> <td>101011</td> <td>-10.5</td> </tr> <tr> <td>010110</td> <td>+11</td> <td>101010</td> <td>-11</td> </tr> <tr> <td>010101</td> <td>+10.5</td> <td>101001</td> <td>-11.5</td> </tr> <tr> <td>010100</td> <td>+10</td> <td>101000</td> <td>-12</td> </tr> <tr> <td></td> <td>etc.</td> <td></td> <td></td> </tr> </tbody> </table>		G (dB)		G (dB)	011000	+12	101100	-10	010111	+11.5	101011	-10.5	010110	+11	101010	-11	010101	+10.5	101001	-11.5	010100	+10	101000	-12		etc.		
	G (dB)		G (dB)																											
011000	+12	101100	-10																											
010111	+11.5	101011	-10.5																											
010110	+11	101010	-11																											
010101	+10.5	101001	-11.5																											
010100	+10	101000	-12																											
	etc.																													

**PRESCALE\_I2S\_0****I2S\_0 Prescaling Register**

Address: 98h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	PRESCALE_I2S_0[5:0]					

Bit Name	Reset	Function																												
Bits [7:6]	00	Reserved.																												
PRESCALE_I2S_0[5:0]	000000	-12 to + 12 dB I2S_0 prescaling to normalize the I2S_0 signal level before audio processing. Auto level control can be implemented by I2C software using the Peak Level Detector. (Default value = 0 dB)																												
		<table border="0"> <thead> <tr> <th></th> <th>G (dB)</th> <th></th> <th>G (dB)</th> </tr> </thead> <tbody> <tr> <td>011000</td> <td>+12</td> <td>101100</td> <td>-10</td> </tr> <tr> <td>010111</td> <td>+11.5</td> <td>101011</td> <td>-10.5</td> </tr> <tr> <td>010110</td> <td>+11</td> <td>101010</td> <td>-11</td> </tr> <tr> <td>010101</td> <td>+10.5</td> <td>101001</td> <td>-11.5</td> </tr> <tr> <td>010100</td> <td>+10</td> <td>101000</td> <td>-12</td> </tr> <tr> <td></td> <td>etc.</td> <td></td> <td></td> </tr> </tbody> </table>		G (dB)		G (dB)	011000	+12	101100	-10	010111	+11.5	101011	-10.5	010110	+11	101010	-11	010101	+10.5	101001	-11.5	010100	+10	101000	-12		etc.		
	G (dB)		G (dB)																											
011000	+12	101100	-10																											
010111	+11.5	101011	-10.5																											
010110	+11	101010	-11																											
010101	+10.5	101001	-11.5																											
010100	+10	101000	-12																											
	etc.																													

**PRESCALE\_I2S\_1****I2S\_1 Prescaling Register**

Address: 99h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	PRESCALE_I2S_1[5:0]					

Bit Name	Reset	Function
Bits [7:6]	00	Reserved.



Bit Name	Reset	Function																												
PRESCALE_I2S_1[5:0]	000000	-12 to + 12 dB I2S_1 prescaling to normalize the I2S_1 signal level before audio processing. Auto level control can be implemented by I2C software using the Peak Level Detector. (Default value = 0 dB)																												
		<table> <thead> <tr> <th></th> <th>G (dB)</th> <th></th> <th>G (dB)</th> </tr> </thead> <tbody> <tr> <td>011000</td> <td>+12</td> <td>101100</td> <td>-10</td> </tr> <tr> <td>010111</td> <td>+11.5</td> <td>101011</td> <td>-10.5</td> </tr> <tr> <td>010110</td> <td>+11</td> <td>101010</td> <td>-11</td> </tr> <tr> <td>010101</td> <td>+10.5</td> <td>101001</td> <td>-11.5</td> </tr> <tr> <td>010100</td> <td>+10</td> <td>101000</td> <td>-12</td> </tr> <tr> <td></td> <td>etc.</td> <td></td> <td></td> </tr> </tbody> </table>		G (dB)		G (dB)	011000	+12	101100	-10	010111	+11.5	101011	-10.5	010110	+11	101010	-11	010101	+10.5	101001	-11.5	010100	+10	101000	-12		etc.		
	G (dB)		G (dB)																											
011000	+12	101100	-10																											
010111	+11.5	101011	-10.5																											
010110	+11	101010	-11																											
010101	+10.5	101001	-11.5																											
010100	+10	101000	-12																											
	etc.																													

**PRESCALE\_I2S\_2****I2S\_2 Prescaling Register**

Address: 9Ah

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	PRESCALE_I2S_2[5:0]					

Bit Name	Reset	Function																												
Bits [7:6]	00	Reserved.																												
PRESCALE_I2S_2[5:0]	000000	-12 to + 12 dB I2S_2 prescaling to normalize the I2S_2 signal level before audio processing. Auto level control can be implemented by I2C software using the Peak Level Detector. (Default value = 0 dB)																												
		<table> <thead> <tr> <th></th> <th>G (dB)</th> <th></th> <th>G (dB)</th> </tr> </thead> <tbody> <tr> <td>011000</td> <td>+12</td> <td>101100</td> <td>-10</td> </tr> <tr> <td>010111</td> <td>+11.5</td> <td>101011</td> <td>-10.5</td> </tr> <tr> <td>010110</td> <td>+11</td> <td>101010</td> <td>-11</td> </tr> <tr> <td>010101</td> <td>+10.5</td> <td>101001</td> <td>-11.5</td> </tr> <tr> <td>010100</td> <td>+10</td> <td>101000</td> <td>-12</td> </tr> <tr> <td></td> <td>etc.</td> <td></td> <td></td> </tr> </tbody> </table>		G (dB)		G (dB)	011000	+12	101100	-10	010111	+11.5	101011	-10.5	010110	+11	101010	-11	010101	+10.5	101001	-11.5	010100	+10	101000	-12		etc.		
	G (dB)		G (dB)																											
011000	+12	101100	-10																											
010111	+11.5	101011	-10.5																											
010110	+11	101010	-11																											
010101	+10.5	101001	-11.5																											
010100	+10	101000	-12																											
	etc.																													

**DEEMPHASIS\_DEMATRIX****Deemphasis-Dematrix Register**

Address: 9Bh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	NICAM_DEMATRIX	NICAM_DEEMPH_BY_PASS	FM_DEMATRIX		FM_DEEMPH_BYPASS	FM_DEEMPH_SW

Bit Name	Reset	Function
Bits [7:6]	00	Reserved.

Bit Name	Reset	Function
NICAM_DEMATRIX	0	Dematrixing for NICAM demodulator input: 00: L=ch0, R=ch1 01: L=ch1, R=ch0
NICAM_DEEMPH_BYPASS	0	0: NICAM deemphasis is not bypassed. 1: NICAM deemphasis is bypassed.
FM_DEMATRIX[3:2]	00	Dematrixing for FM demodulator input: 00: L=ch0, R=ch1 01: L=ch0+ch1, R=ch0-ch1 10: L=2ch0-ch1, R=ch1 11: L=(ch0+ch1)/2, R=(ch0-ch1)/2
FM_DEEMPH_BYPASS	0	0: FM deemphasis is not bypassed. 1: FM deemphasis is bypassed.
FM_DEEMPH_SW	0	0: 50 $\mu$ s FM deemphasis. 1: 75 $\mu$ s FM deemphasis.

**PEAK\_DET\_INPUT****Peak Detector Input source Register**

Address: 9Dh

Type: R

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
PEAK_LOCATION	0	PEAK_L_R_RANGE			PEAK_DET_INPUT[1:0]		

Bit Name	Reset	Function
PEAK_LOCATION	0	Peak detector location : 0: Peak detector placed between FM/NICAM Dematrix and Audio Matrix or between I <sup>2</sup> S Prescale and DownMix 1: Peak detector placed before DC removal (For input saturation detection)
Bit 6	0	Reserved.
PEAK_L_R_RANGE	0000	Peak L-R range. 0000 : 0 dBFS to -42 dBFS 0001 : -6 dBFS to -48 dBFS 0010 : -12 dBFS to -54 dBFS 0011 : -18 dBFS to -60 dBFS ...
PEAK_DET_INPUT[1:0]	00	<b>Peak Level Detector Source Selection</b> 00: AM/FM or I2S 0      10: SCART or I2S 2 01: NICAM or I2S 1

**PEAK\_DET\_L****Peak Level Detector Status Register (L channel)**

Address: 9Eh

Type: R

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
OVERLOAD_L	PEAK_L[6:0]						

Bit Name	Reset	Function
OVERLOAD_L[7]	0	Memorise overload on the peak detection. This field can be reset.
PEAK_L[6:0]	00000000	<p>Displays the <b>Absolute Peak Level</b> of the audio source selected. The measured value is updated continuously every 64 ms. The range varies linearly from the full scale (0 dB) down to 1/256 of the full scale (-48 dB).</p> <p>In AM/FM Mono mode, only the PEAK_L[7:0] value must be taken into account.</p> <p>In FM Mono mode, the audio peak level range depends upon the programmed FM bandwidth. The unique difference is that the measurement is done after Sound pre-processing (DC offset removal, Prescaling, De-emphasis and Dematrixing).</p> <p>In FM Stereo mode, the maximum value may be used to check if the incoming signal level is correctly adjusted by the prescaling factor or if there are no FM overmodulation problems (clipping).</p> <p>Programmable values are listed in <a href="#">Table 21</a>.</p>

**PEAK\_DET\_R****Peak Level Detector Status Register (R channel)**

Address: 9Fh

Type: R

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
OVERLOAD_R	PEAK_R[6:0]						

Bit Name	Reset	Function
OVERLOAD_R[7]	0	Memorise overload on the peak detection. This field can be reset.
PEAK_R[7:0]	00000000	<p>Displays the <b>Absolute Peak Level</b> of the audio source selected. The measured value is updated continuously every 64 ms. The range varies linearly from the full scale (0 dB) down to 1/256 of the full scale (-48 dB).</p> <p>For more information, refer to register <a href="#">PEAK_DET_L</a>.</p>

**PEAK\_DET\_L\_R****Peak Level Detector Status Register (L - R)**

Address: A0h

Type: R

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
OVERLOAD_L_R	PEAK_L_R[6:0]						

Bit Name	Reset	Function
OVERLOAD_L_R[7]	0	Memorise overload on the peak detection. This field can be reset.
PEAK_L_R[7:0]	00000000	<p>Displays the <b>Difference between L and R (L - R) channels</b> for the audio source selected.</p> <p>For more information, refer to register <a href="#">PEAK_DET_L</a>.</p>

## 12.14 Matrixing

**AUDIO\_MATRIX\_INPUT****Audio Matrix Input Selection Register**

Address: A2h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	SCART_ INPUT_ SOURCE	HP_INPUT_ SOURCE	LS_INPUT_ SOURCE

Bit Name	Reset	Function
Bits [7:3]	00000	Reserved.
SCART_INPUT_ SOURCE	0	Select input source for SCART output: 0: Demod 1: SCART input
HP_INPUT_ SOURCE	0	Select input source for HP output: 0: Demod 1: SCART input
LS_INPUT_ SOURCE	0	Select input source for LS output: 0: Demod 1: SCART input

**AUDIO\_MATRIX\_CONFIG**

Address: A3h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	SCART_ MATRIX	DEMOD_MATRIX[3:0]			

Bit Name	Reset	Function
Bits [7:5]	000	Reserved.
SCART_MATRIX	0	Indicates the SCART input signal matrixing (see <a href="#">Table 23</a> )
DEMOD_MATRIX [3:0]	0000	Indicates the demod input signal matrixing (see <a href="#">Table 22</a> )

Table 22: Demod Matrix

Input Mode	Language -> demod_mx	Stereo		Mono A		Mono B		Mono C		Backup mode
		L	R	L	R	L	R	L	R	
Mono AM/FM with backup	0000	FM		FM		FM		FM		
Mono AM/FM no backup	0001	-		-		-		FM		
Zwt St	0100	FM_L	FM_R	$(FM\_L + FM\_R)/2$		$(FM\_L + FM\_R)/2$		$(FM\_L + FM\_R)/2$		
Zwt Dual	0101	FM_M1	FM_M2	FM_M1		FM_M2		$(FM\_M1 + FM\_M2)/2$		
NICAM Mn, backup	1000	NIC_M1		NIC_M1		NIC_M1		FM		Mono AM/FM with backup
NICAM Dual backup	1001	NIC_M1	NIC_M2	NIC_M1		NIC_M2		FM		Mono AM/FM with backup
NICAM St, backup	1010	NIC_L	NIC_R	$(NIC\_L + NIC\_R)/2$		$(NIC\_L + NIC\_R)/2$		FM		Mono AM/FM with backup
NICAM Mn, no backup	1100	NIC_M1		NIC_M1		NIC_M1		FM		Mono AM/FM no backup
NICAM Dual, no backup	1101	NIC_M1	NIC_M2	NIC_M1		NIC_M2		FM		Mono AM/FM no backup
NICAM St, no backup	1110	NIC_L	NIC_R	$(NIC\_L + NIC\_R)/2$		$(NIC\_L + NIC\_R)/2$		FM		Mono AM/FM no backup

Note: Switching between Stereo and Forced Mono modes can be done using  $(FM\_L + FM\_R)/2$  or  $(NIC\_L + NIC\_R)/2$  configurations.

Table 23: SCART Matrix

SCART_MX	Stereo		Mono A		Mono B		Mono C	
	Left	Right	Left	Right	Left	Right	Left	Right
0	SCART_L	SCART_R	SCART_L		SCART_R		$(SCART\_L + SCART\_R)/2$	
1	SCART_R	SCART_L	SCART_R		SCART_L		$(SCART\_L + SCART\_R)/2$	

## AUDIO\_MATRIX\_LANGUAGE

Address: A4h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
MUTE_STEREO	MUTE_ALL	SCART_LANGUAGE[1:0]		HP_LANGUAGE[1:0]		LS_LANGUAGE[1:0]	

Bit Name	Reset	Function
MUTE_STEREO	0	Mute outputs with stereo signal input
MUTE_ALL	0	Mute all outputs
SCART_LANGUAGE[1:0]	00	Select language for SCART output
HP_LANGUAGE[1:0]	00	Select language for HPoutput
LS_LANGUAGE[1:0]	00	Select language for LS output 00: stereo 01: mono A 10: mono B 11: mono C

### DOWNMIX\_IN\_MODE

Address: A6h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	LFE_IN	MIX_IN_MODE[2:0]		

Bit Name	Reset	Function
Bits[7:4]	0000	Reserved
LFE_IN	0	0: LFE signal is not inputed throught Downmix Block 1: LFE signal is inputed through Downmix Block
MIX_IN_MODE[2:0]	010	see <a href="#">Table 24</a>

**Table 24: DownMix IN modes**

Parameter Coding (Decimal Format)	Parameter Field Label	Function
0	MODE11	Mode not used in STV82x7
1	MODE10	1/0 (C)
2	MODE20	2/0 (L,R)
3	MODE30	3/0 (L,R,C)
4	MODE21	2/1 (L,R,S)
5	MODE31	3/1 (L,R,C,S)
6	MODE22	2/2 (L,R,Ls,Rs)
7	MODE32	3/2 (L,R,C,Ls,Rs)

**DOWNMIX\_OUT\_MODE**

Address:A7h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	HP_MODE[1:0]		SCART_MODE[1:0]		LS_OUT_MODE[2:0]		

Bit Name	Reset	Function
Bit 7	0	Reserved.
HP_MODE[1:0]	10	see <a href="#">Table 25</a>
SCART_MODE[1:0]	01	see <a href="#">Table 25</a>
LS_OUT_MODE [2:0]	010	see <a href="#">Table 26</a>

**Table 25: DownMix SCART/HP modes**

Parameter Coding (Decimal Format)	Parameter Field Label	Function
0	MIX_VCR_OFF	Switch off the VCR table setup
1	MIX_VCR_PROLOGIC	VCR table setup for Tape outputs (for later decoding by a Dolby Prologic decoder - Lt,Rt)
2	MIX_VCR_STEREO	VCR table setup for Stereo and headphone listening (Lo,Ro)
3	MIX_COSTOM	reserved

**Table 26: DownMix LS OUT modes**

Parameter Coding (Decimal Format)	Parameter Field Label	Function
0	MODE20t	2/0 Dolby Surround (Lt,Rt)
1	MODE10	1/0 (C)
2	MODE20	2/0 (L,R)
3	MODE30	3/0 (L,R,C)
4	MODE21	2/1 (L,R,S)
5	MODE31	3/1 (L,R,C,S)
6	MODE22	2/2 (L,R,Ls,Rs)
7	MODE32	3/2 (L,R,C,Ls,Rs)

**DOWNMIX\_DUAL\_MODE**

Address: A8h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	DUAL_ON	LS_DUAL_SELECT[1:0]	SCART_DUAL_SELECT[1:0]	HP_DUAL_SELECT[1:0]			

Bit Name	Reset	Function
Bit 7	0	Reserved.
DUAL_ON	0	0: Dual mode disable 1: Dual mode enable
LS_DUAL_SELECT[1:0]	00	Dual Mono Mode on LS output 00: LS dual stereo                    10: LS dual right mono 01: LS dual left mono                11: LS dual mixed
SCART_DUAL_SELECT[1:0]	00	Dual Mono Mode on SCART output 00: SCART dual stereo                10: SCART dual right mono 01: SCART dual left mono            11: SCART dual mixed
HP_DUAL_SELECT[1:0]	00	Dual Mono Mode on HP output 00: HP dual stereo                    10: HP dual right mono 01: HP dual left mono                11: HP dual mixed

**DOWNMIX\_CONFIG**

Address: A9h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	SRND_FACTOR[1:0]	CENTER_FACTOR[1:0]	LR_UPMIX	NORMALIZE		

Bit Name	Reset	Function
Bits[7:6]	00	
SRND_FACTOR [1:0]	00	00: -3 dB                                10: -6 dB 01: -4.5 dB                              11: -6 dB
CENTER_FACTOR [1:0]	00	00: -3 dB                                10: -6 dB 01: -4.5 dB                              11: -4.5 dB
LR_UPMIX	0	0: Disable upmixing 1: Enable upmixing (DTS specified)
NORMALIZE	1	0: Disable normalization 1: Enable normalization



## 12.15 Audio Processing

## PRO\_LOGIC2\_CONTROL

Address: AAh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
PL2_LFE	PL2_OUTPUT_DOWNMIX[2:0]			PL2_MODES[2:0]			PL2_ACTIVE

Bit Name	Reset	Function
PL2_LFE	0	0: Reset the LFE channel 1: Bypass the LFE channel
PL2_OUTPUT_DOWNMIX[2:0]	000	000: not applicable 001: not applicable 010: not applicable 011: 3/0 output mode (L,R,C) 100: 2/1 output mode (L,R,Ls - phantom) 101: 3/1 output mode (L,R,C,Ls) 110: 2/2 output mode (L,R,Ls,Rs - phantom) 111: 3/2 output mode (L,R,C,Ls,Rs)
PL2_MODES[2:0]	000	000: Pro Logic 1 Emulation (forced if DPL version) 001: Virtual (DPL2 version only) 010: Music (DPL2 version only) 011: Movie (standard) (DPL2 version only) 100: Matrix (DPL2 version only) 101: Custom (DPL2 version only) 110: not applicable (DPL2 version only) 111: not applicable (DPL2 version only)
PL2_ACTIVE	0	0: Dolby Prologic 2 is not active 1: Dolby Prologic 2 is active

Table 27: Prologic II Decode Mode Configuration

PL2 Mode	Decode Mode	Dimension	Center Width	Auto-Balance	Panorama	Surround Coherence	SUR Filtering
0	Pro Logic Emulation	3	0	1	0	0	2
1	Virtual	3	0	1	0	1	0
2	Music	x	x	0	x	1	1
3	Movie/ Standard	3	0	1	0	0	0
4	Matrix	3	0	0	0	1	1
5	Custom	x	x	x	x	x	x

Note: (x = user defined parameter)

### PCM\_SRND\_DELAY

Address: ABh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	SNRD_DELAY[4:0]				

Bit Name	Reset	Function
Bits[7:5]	000	Reserved.
SNRD_DELAY[4:0]	00000	Surround Channel Delay range: 0 to 30 (in ms)

Note: See [Table 19](#) for audio/video delay configuration.

### PCM\_CENTER\_DELAY

Address: ACh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	CENTER_DELAY[3:0]			

Bit Name	Reset	Function
Bits[7:4]	0000	Reserved.
CENTER_DELAY[3:0]	0000	Center Channel Delay range: 0 to 10 (in ms)

Note: See [Table 19](#) for audio/video delay configuration.

### PRO\_LOGIC2\_CONFIG

Address: ADh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
PL2_LFE	0	0	PL2_SRND_FILTER[1:0]	PL2_RS_POLARITY	PL2_PANORAMA	PL2_AUTOBALANCE	

Bit Name	Reset	Function
Bits[7:6]	00	Reserved.
PL2_SRND_FILTR[1:0]	00	00: 0: Off 01: 1: Shelf Filter (for music and matrix modes) 10: 2: 7 kHz LP 11: 3: not applicable
PL2_RS_POLARITY	0	0: Rs polarity normal 1: Rs polarity inverted
PL2_PANORAMA	0	0: Panorama Off 1: Panorama On
PL2_AUTOBALANCE	0	0: Autobalance Off 1: Autobalance On

See [Table 27: Prologic II Decode Mode Configuration](#) for programming of these bits depending on the decode mode.

### PRO\_LOGIC2\_DIMENSION

Address: AEh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	PL2_C_WIDTH			0	PL2_DIMENSION		

Bit Name	Reset	Function
Bit 7	0	Reserved.
PL2_C_WIDTH[2:0]	000	000: 0, No Spread = OFF      100: 54 001: 20                              101: 62 010: 28                              110: 69 011: 36                              111: 90, Phantom
Bit 3	0	Reserved.
PL2_DIMENSION[2:0]	000	000: -3, most surround 001: -2 010: -1 011: 0, neutral = OFF 100: 1 101: 2 110: 3, most center 111: not applicable

See [Table 27: Prologic II Decode Mode Configuration](#) for programming of these bits depending on the decode mode.

**PRO\_LOGIC2\_LEVEL**

Address: AFh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
PL2_LEVEL							

Bit Name	Reset	Function
PL2_LEVEL[7:0]	00000000	Input Gain attenuation: 0000 0000: 0 dB 0000 0001: -0.5 dB ... 1111 1111: -127.5 dB

**NOISE\_GENERATOR**

Address: B0h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
10_DB_ ATTENUATE	SRIGHT_ NOISE	SLEFT_ NOISE	SUB_ NOISE	CENTER_ NOISE	RIGHT_ NOISE	LEFT_ NOISE	NOISE_ON

Bit Name	Reset	Function
10_DB_ATTENUATE	0	0: noise is outputed with full range 1: noise is outputed with a 10 dB attenuation
SRIGHT_NOISE	0	1: Generates noise on LS right surround output
SLEFT_NOISE	0	1: Generates noise on LS left surround output
SUB_NOISE	0	1: Generates noise on LS subwoofer output
CENTER_NOISE	0	1: Generates noise on LS center output
RIGHT_NOISE	0	1: Generates noise on LS right output
LEFT_NOISE	0	1: Generates noise on LS left output
NOISE_ON	0	0: Noise Generation not active 1: Noise Generation is active

**TRUSRND\_CONTROL**

Address: B1h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	TRUSRND_ MONO_SRND	TRUSRND_INPUT_MODE[3:0]				TRUSRND_ MODE	TRUSRND_ ON

Bit Name	Reset	Function
Bit 7	0	Reserved.
TRUSRND_MONO_SRND	0	0: Left mono Srnd mode 1: Right mono Srnd mode
TRUSRND_INPUT_MODE[3:0]	0000	0000: Mono 0001: L/R stereo (SRS mode) 0010: L/R/S (SRS mode, Prologic 1 Process) 0011: L/R/Ls/Rs (SRS mode) 0100: L/R/C (TruSurround mode) 0101: L/R/C/S (TruSurround mode, Prologic 1 Process) 0110: L/R/C/Ls/Rs (TruSurround mode) 0111: Lt/Rt (TruSurround mode) 1000: L/R/C/Ls/Rs (SRS mode, BS Digital Broadcast) 1001: L/R/C/Ls/Rs (TruSurround, Prologic 2 Music mode)
TRUSRND_MODE	0	0: TruSurround mode 1: Bypass mode
TRUSRND_ON	0	0: TruSurround OFF 1: TruSurround ON

**Note:** How to use TruSurround XT:

- Implementation of TruSurround XT is done by setting the TRUSRND\_ON bit to 1.
- TruSurround XT mode must be selected by TRUSRND\_INPUT\_MODE[3:0] bits.
- Activation or non-activation of TruSurround XT must be done by using the TRUSRND\_MODE bit.

### TRUSRND\_INPUT\_GAIN

Address: B6h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
TRUSRND_INPUT_GAIN[7:0]							

Bit Name	Reset	Function
TRUSRND_INPUT_GAIN[7:0]	0000 0000	Input Gain attenuation: 0000 0000: 0 dB 0000 0001: -0.5 dB ... 1111 1111: -127.5 dB

### TRUSRND\_HP\_DCL

Address: B7h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	DIALOG_CLARITY_ON	HEADPHONE_ON	0

Bit Name	Reset	Function
Bits[7:2]	00000	Reserved.
DIALOG_CLARITY_ON	0	0: Dialog Clarity OFF 1: Dialog Clarity ON
HEADPHONE_ON	0	Activate HP mode in TruSurround XT: 0: HP mode OFF 1: HP mode ON
Bit [0]	0	Reserved.

### TRUSRND\_DC\_ELEVATION

Address: B8h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
TRUSRND_DC_ELEVATION[7:0]							

Bit Name	Reset	Function
TRUSRND_DC_ELEVATION[7:0]	0000 1100	Dialog Clarity Elevation: 0000 0000: 0 dB 0000 0001: -0.5 dB ... 1111 1111: -127.5 dB

### TRUBASS\_LS\_CONTROL

Address: BAh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	TRUBASS_LS_SIZE[2:0]		TRUBASS_LS_ON	

Bit Name	Reset	Function
Bits[7:3]	00000	Reserved.
TRUBASS_LS_SIZE[2:0]	011	000: LF response at 40 Hz 001: LF response at 60 Hz 010: LF response at 100 Hz 011: LF response at 150 Hz 100: LF response at 200 Hz 101: LF response at 250 Hz 110: LF response at 300 Hz 111: LF response at 400 Hz
TRUBASS_LS_ON	0	0: LS TruBass OFF 1: LS TruBass ON

**TRUBASS\_LS\_LEVEL**

Address: BBh

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

TRUBASS_LS_LEVEL[7:0]
-----------------------

Bit Name	Reset	Function
TRUBASS_LS_LEVEL[7:0]	0000 1001	Define the amount of SRS TruBass effect for LS outputs: 0000 0000: 0 dB 0000 0001: -0.5 dB ... 1111 1111: -127.5 dB

**TRUBASS\_HP\_CONTROL**

Address: BCh

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

0	0	0	0	TRUBASS_HP_SIZE[2:0]	TRUBASS_HP_ON
---	---	---	---	----------------------	---------------

Bit Name	Reset	Function
Bits[7:3]	00000	Reserved.
TRUBASS_HP_SIZE[2:0]	011	000: LF response at 40 Hz      100: LF response at 200 Hz 001: LF response at 60 Hz      101: LF response at 250 Hz 010: LF response at 100 Hz      110: LF response at 300 Hz 011: LF response at 150 Hz      111: LF response at 400 Hz
TRUBASS_HP_ON	0	0: HP TruBass OFF 1: HP TruBass ON

**TRUBASS\_HP\_LEVEL**

Address: BDh

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

TRUBASS_HP_LEVEL[7:0]
-----------------------

Bit Name	Reset	Function
TRUBASS_HP_LEVEL[7:0]	0000 1001	Define the amount of SRS TruBass effect for HP outputs: 0000 0000: 0 dB 0000 0001: -0.5 dB ... 1111 1111: -127.5 dB

### SVC\_LS\_CONTROL

Address: BEh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	SVC_LS_INPUT[1:0]		SVC_LS_AMP	SVC_LS_ON

Bit Name	Reset	Function
Bits[7:4]	0000	Reserved.
SVC_LS_INPUT[1:0]	00	Select input for peak detection in multichannel mode: 00: Left/Right 01: Center 10: Left/Right/Center
SVC_LS_AMP	1	0: 0 dB amplification in auto-mode 1: +6 dB amplification in auto-mode
SVC_LS_ON	0	0: Manual mode(simple prescaler) 1: Automatic mode

### SVC\_LS\_TIME\_TH

Address: BFh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
SVC_LS_TIME[2:0]				SVC_LS_THRESHOLD[4:0] (S)			

Bit Name	Reset	Function
SVC_LS_TIME[2:0]	100	Time constant for the amplification (6 dB gain step) in automatic mode: 000: 30 ms                      100: 16 s 001: 200 ms                    101: 32 s 010: 500 ms                    110: 64 s 011: 1 s                         111: 128 s
SVC_LS_THRESHOLD[4:0]	11000	See <a href="#">Table 28</a> and <a href="#">Table 29</a> .



Table 28: Gain (threshold field) values in Manual mode

Manual Mode	Gain (dB)	Manual Mode	Gain (dB)
00101	+15.5	11101	-8.5
00100	+12	11100	-12
00011	+9.5	11011	-14.5
00010	+6	11010	-18
00001	+3.5	11001	-20.5
00000	0	11000	-24
11111	-2.5	10111	-26.5
11110	-6	10110	-30

Table 29: Threshold values in Automatic mode

Automatic Mode	Threshold (dB)	Automatic Mode	Threshold (dB)
11111	-2.5	11010	-18
11110	-6	11001	-20.5
11101	-8.5	11000	-24
11100	-12	10111	-26.5
11011	-14.5	10110	-30

## SVC\_HP\_CONTROL

Address: C0h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	0	SVC_LHP_AMP	SVC_HP_ON

Bit Name	Reset	Function
Bits[7:2]	000000	Reserved.
SVC_LHP_AMP	1	0: 0 dB amplification in auto-mode 1: +6 dB amplification in auto-mode
SVC_HP_ON	0	0: Manual mode (simple prescaler) 1: Automatic mode

**SVC\_HP\_TIME\_TH**

Address: C1h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
SVC_HP_TIME[2:0]				SVC_HP_THRESHOLD[4:0] (S)			

Bit Name	Reset	Function
SVC_HP_TIME[2:0]	100	Time constant for the amplification (6 dB gain step) in automatic mode: 000: 30 ms                      100: 16 s 001: 200 ms                    101: 32 s 010: 500 ms                    110: 64 s 011: 1 s                         111: 128 s
SVC_HP_THRESHOLD[4:0]	11000	see <a href="#">Table 28</a> and <a href="#">Table 29</a>

**SVC\_LS\_GAIN**

Address: C2h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	SVC_LS_GAIN[6:0]						

Bit Name	Reset	Function
Bit 7	0	Reserved.
SVC_LS_GAIN[6:0]	0000000	Set "make-up" gain applied at SVC LS output: 0000000: +0 dB 0000001: +0.5 dB ... 0101110: +23 dB 0101111: +23.5 dB 0110000: +24 dB

**SVC\_HP\_GAIN**

Address: C3h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	SVC_HP_GAIN[6:0]						

Bit Name	Reset	Function
Bit 7	0	Reserved.

Bit Name	Reset	Function
SVC_HP_GAIN[6:0]	0000000	Set "make-up" gain applied at SVC HP output: 0000000: +0 dB 0000001: +0.5 dB ... 0101110: +23 dB 0101111: +23.5 dB 0110000: +24 dB

**STSRND\_CONTROL****ST WideSurround Control Register**

Address: C4h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	STSRND_STEREO	STSRND_MODE	STSRND_ON

Bit Name	Reset	Function
Bits[7:3]	00000	Reserved.
STSRND_STEREO	0	<b>ST WideSurround Mode</b> 0: ST WideSurround Sound in Mono mode (Default) 1: ST WideSurround Sound in Stereo mode
STSRND_MODE	0	<b>ST WideSurround Sound Stereo Mode</b> 0: Movie Mode 1: Music Mode
STSRND_ON	0	<b>ST WideSurround Sound Enable</b> 0: ST WideSurround Sound is disabled 1: ST WideSurround Sound is enabled

**STSRND\_FREQ****ST WideSurround Sound Frequency**

Address: C5h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	STSRND_BASS[1:0]		STSRND_MEDIUM[1:0]		STSRND_TREBLE[1:0]	

Bit Name	Reset	Function
Bits[7:6]	00	Reserved.
STSRND_BASS[1:0]	01	Defines the bass frequency effect for ST WideSurround Sound. Programmable values are listed in <a href="#">Table 30</a> .
STSRND_MEDIUM[1:0]	01	Defines the medium frequency effect for ST WideSurround Sound in Movie or Mono mode (no effect in Music mode). Programmable values are listed in <a href="#">Table 30</a> .

Bit Name	Reset	Function
STSRND_TREBLE[1:0]	01	Defines the treble frequency effect for ST WideSurround Sound in Movie or Mono mode (no effect in Music mode). Programmable values are listed in <a href="#">Table 30</a> .

Table 30: Phase Shifter Center Frequencies

	Phase Shifter Center Frequency		
	BASS_FREQ[1:0]	MEDIUM_FREQ[1:0]	TREBLE_FREQ[1:0]
00	40 Hz	202 Hz	2 kHz
01 (Default)	90 Hz	416 Hz	4 kHz
10	120 Hz	500 Hz	5 kHz
11	160 Hz	588 Hz	6 kHz

**STSRND\_LEVEL****ST WideSurround Gain Register**

Address: C6h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
STSRND_GAIN[7:0]							

Bit Name	Reset	Function																								
STSRND_GAIN[7:0]	10000000	Defines the ST WideSurround Sound component gain in linear scale. <table border="1" style="margin-left: 20px; width: 100%;"> <thead> <tr> <th></th> <th>Level (%)</th> <th></th> <th>Level (%)</th> </tr> </thead> <tbody> <tr> <td>1000 0000 (Default)</td> <td>100%</td> <td>0000 0100</td> <td>3.1%</td> </tr> <tr> <td>0111 1111</td> <td>99.2%</td> <td>0000 0011</td> <td>2.3%</td> </tr> <tr> <td>0111 1110</td> <td>98.4%</td> <td>0000 0010</td> <td>1.6%</td> </tr> <tr> <td>0111 1101</td> <td>97.6%</td> <td>0000 0001</td> <td>0.8%</td> </tr> <tr> <td>.....</td> <td></td> <td>0000 0000</td> <td>0%</td> </tr> </tbody> </table>		Level (%)		Level (%)	1000 0000 (Default)	100%	0000 0100	3.1%	0111 1111	99.2%	0000 0011	2.3%	0111 1110	98.4%	0000 0010	1.6%	0111 1101	97.6%	0000 0001	0.8%	.....		0000 0000	0%
	Level (%)		Level (%)																							
1000 0000 (Default)	100%	0000 0100	3.1%																							
0111 1111	99.2%	0000 0011	2.3%																							
0111 1110	98.4%	0000 0010	1.6%																							
0111 1101	97.6%	0000 0001	0.8%																							
.....		0000 0000	0%																							

**OMNISURROUND\_CONTROL**

Address: C7h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
LFE	ST_VOICE[1:0]		FRONT_BYPASS	OMNI_SURND_INPUT_MODE[3:0]		OMNISURND_ON	

Bit Name	Reset	Function
LFE	0	0: Do not use LFE channel 1: Generate LFE channel

Bit Name	Reset	Function	
ST_VOICE[1:0]	00	00: OFF 01: Low	10: Mid 11: High
FRONT_BYPASS	0	Forced to 0	
OMNISRND_ INPUT_MODE[3:0]	0000	000: Mono 001: L/R stereo 010: L/R/S 011: L/R/Ls/Rs	100: L/R/C 101: L/R/C/S 110: L/R/C/Ls/Rs 111: Lt/Rt (Passive matrix)
OMNISURND_ON	0	0: OmniSurround OFF 1: OmniSurround ON	

### ST\_DYNAMIC\_BASS

Address: C8h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
BASS_LEVEL[4:0]					BASS_FREQ[1:0]		DYN_BASS_ON

Bit Name	Reset	Function	
BASS_LEVEL[4:0]	00000	Set ST Dynamic Bass effect level: 00000: +0 dB 00001: +0.5 dB ... 11101: +14.5 dB 11110: +15 dB 11111: +15.5 dB	
BASS_FREQ[1:0]	00	00: 100 Hz Cut-Off frequency 01: 150 Hz Cut-Off frequency 10: 200 Hz Cut-Off frequency 11: Reserved	
DYN_BASS_ON	0	0: ST Dynamic Bass OFF 1: ST Dynamic Bass ON	

## 12.16 5-Band Equalizer / Bass-Treble for Loudspeakers

### LS\_EQ\_BT\_CTRL

### Loudspeakers Equalizer Control Register

Address: C9h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	0	LS_EQ_BT_SW	LS_EQ_ON

Bit Name	Reset	Function
Bits[7:2]	000000	Reserved.
LS_EQ_BT_SW	0	<b>5-Band Equalizer or Bass-Treble selection</b> 0: 5-Band Equalizer is selected for Loudspeakers. 1: Bass-Treble is selected for Loudspeakers.
LS_EQ_ON	1	<b>5-Band Equalizer/Bass-Treble for loudspeakers Enable</b> 0: 5-Band Equalizer/Bass-Treble is disabled 1: 5-Band Equalizer/Bass-Treble is enabled (Default)

**EQ\_BANDX\_GAIN****Loudspeakers Equalizer Gain Register for BandX**

Address: CAh to CEh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
EQ_BANDX							

Bit Name	Reset	Function
EQ_BANDX[7:0]	0000 0000	BandX gain adjustment within a range from -12 dB to +12 dB in steps of 0.25 dB. Band1: 100 Hz, Band2: 330 Hz, Band3: 1 kHz, Band4: 3.3 kHz, Band5: 10 kHz, see <a href="#">Table 31</a> .

*Note:* With positive equalizer settings, internal clipping may occur even with an overall volume of less than 0 dB. This will lead to a clipped output signal. Therefore, it is not recommended to set equalizer bands to a value that, in conjunction with volume, would result in an overall positive gain.

**Table 31: Loudspeakers Equalizer/Bass-Treble Gain Values (and Headphone Bass-Treble Gain Values)**

Value	Gain G (dB)
00110000	+12
00101111	+11.75
00101110	+11.50
.....	.....
00000000 (Default)	0
.....	.....
11010010	-11.50
11010001	-11.75
11010000	-12

**LS\_BASS\_GAIN****Loudspeakers Bass Gain Register**

Address: CFh

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

LS\_BASS[7:0]

Bit Name	Reset	Function
LS_BASS[7:0]	0000 0000	Bass gain adjustment within a range from -12 dB to +12 dB in steps of 0.25 dB.

*Note:* With positive bass/treble settings, internal clipping may occur even with an overall volume of less than 0 dB. This will lead to a clipped output signal. Therefore, it is not recommended to set bass/treble bands to a value that, in conjunction with volume, would result in an overall positive gain.

**LS\_TREBLE\_GAIN****Loudspeakers Treble Gain Register**

Address: D0h

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

LS\_TREBLE

Bit Name	Reset	Function
LS_TREBLE[7:0]	0000 0000	Treble gain adjustment within a range from -12 dB to +12 dB in steps of 0.25 dB.

*Note:* With positive bass/treble settings, internal clipping may occur even with an overall volume of less than 0 dB. This will lead to a clipped output signal. Therefore, it is not recommended to set bass/treble bands to a value that, in conjunction with volume, would result in an overall positive gain.

**12.17 Headphone Bass-Treble****HP\_BT\_CONTROL****Headphone Bass-Treble Control Register**

Address: D1h

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

0	0	0	0	0	0	0	HP_BT_ON
---	---	---	---	---	---	---	----------

Bit Name	Reset	Function
Bits [7:1]	0000000	Reserved.
HP_EQ_ON	1	<b>Bass-Treble for headphone Enable</b> 0: Bass-Treble is disabled 1: Bass-Treble is enabled (Default)

**HP\_BASS\_GAIN****Headphone Bass Gain**

Address: D2h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
HP_BASS_GAIN[7:0]							

Bit Name	Reset	Function
HP_BASS_GAIN[7:0]	00000000	<b>Gain Tuning of Headphone Bass Frequency</b> Gain may be programmed within a range between +12 dB and -12 dB in steps of 0.25 dB. Programmable values are listed in <a href="#">Table 31</a> .

*Note:* With positive bass/treble settings, internal clipping may occur even with an overall volume of less than 0 dB. This will lead to a clipped output signal. Therefore, it is not recommended to set bass/treble bands to a value that, in conjunction with volume, would result in an overall positive gain.

**HP\_TREBLE\_GAIN****Headphone Treble Gain**

Address: D3h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
HP_TREBLE_GAIN[4:0]							

Bit Name	Reset	Function
HP_TREBLE_GAIN[7:0]	00000000	<b>Gain Tuning of Headphone Treble Frequency</b> Gain may be programmed within a range between +12 dB and -12 dB in steps of 0.25 dB. Programmable values are listed in <a href="#">Table 31</a> .

*Note:* With positive bass/treble settings, internal clipping may occur even with an overall volume of less than 0 dB. This will lead to a clipped output signal. Therefore, it is not recommended to set bass/treble bands to a value that, in conjunction with volume, would result in an overall positive gain.







Bit Name	Reset	Function
ANTICLIP_LS_VOL_CLAMP	1	The output level is clamped depending on the LS Equalizer or LS Bass-Treble value to avoid any possible signal clipping on LS output. 0: Volume clamp on LS output is not active 1: Volume clamp on LS output is active
Bits[5:4]	00	Reserved.
SCART_VOLUME_MODE	0	Volume mode for SCART output: 0: independent 1: Differential
SRND_VOLUME_MODE	1	Volume mode for Headphone output: 0: independent 1: Differential
HP_VOLUME_MODE	1	Volume mode for Surround output: 0: independent 1: Differential
LS_VOLUME_MODE	1	Volume mode for LS output: 0: independent 1: Differential

Note: 1 For the use of volume and balance control please refer to [Figure 20](#) and [Figure 21](#).

2 In differential mode the left register is used for volume control and the right register is used for balance control.

### LS\_L\_VOLUME\_MSB

Address: D8h

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

LS\_L\_VOLUME\_MSB[7:0]

Bit Name	Reset	Function
LS_L_VOLUME_MSB[7:0]	1001 1000	LS 10 bits volume Left channel 8 MSB in independent mode or LS 10 bits volume Left and Right channels 8 MSB in differential mode. See <a href="#">Figure 20: Volume Control on page 38</a> for range values.

### LS\_L\_VOLUME\_LSB

Address: D9h

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

0      0      0      0      0      0      LS\_L\_VOLUME\_LSB[1:0]

Bit Name	Reset	Function
Bits[7:2]	000000	Reserved.
LS_L_VOLUME_LSB[1:0]	00	LS 10 bits volume Left channel 2 LSB in independent mode or LS 10 bits volume Left and Right channels 2 LSB in differential mode. See <a href="#">Figure 20: Volume Control on page 38</a> for range values.

The volume value is defined by the following formula:

$\text{Vol (dB)} = \text{Decimal value of LS\_L\_VOLUME\_MSB} \times 0.5 + \text{Decimal value of LS\_L\_VOLUME\_LSB} \times 0.125 - 116 \text{ dB}$   
(each step is 0.125 dB).

### LS\_R\_VOLUME\_MSB

Address: DAh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
LS_R_VOLUME_MSB[7:0]							

Bit Name	Reset	Function
LS_R_VOLUME_MSB[7:0]	0000000 0	LS 10 bits volume Right channel 8 MSB in independent mode or LS 10 bits Left and Right balance 8 MSB in differential mode. See <a href="#">Figure 20: Volume Control on page 38</a> or <a href="#">Figure 21: Differential Balance on page 39</a> .

### LS\_R\_VOLUME\_LSB

Address: DBh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	0	LS_R_VOLUME_LSB[1:0]	

Bit Name	Reset	Function
Bits[7:2]	000000	Reserved.
LS_R_VOLUME_LSB[1:0]	00	LS 10 bits volume Right channel 2 LSB in independent mode or LS 10 bits Left and Right balance 2 LSB in differential mode. See <a href="#">Figure 20: Volume Control on page 38</a> or <a href="#">Figure 21: Differential Balance on page 39</a> .

**LS\_C\_VOLUME\_MSB**

Address: DCh

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

LS_C_VOLUME_MSB[7:0]
----------------------

Bit Name	Reset	Function
LS_C_VOLUME_MSB[7:0]	1001 1000	LS 10 bits volume Center channel 8 MSB See <a href="#">Figure 20: Volume Control on page 38</a> for range values.

**LS\_C\_VOLUME\_LSB**

Address: DDh

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

0	0	0	0	0	0	0	LS_C_VOLUME_LSB[1:0]
---	---	---	---	---	---	---	----------------------

Bit Name	Reset	Function
Bits[7:2]	000000	Reserved.
LS_C_VOLUME_LSB[1:0]	00	LS 10 bits volume Center channel 2 LSB See <a href="#">Figure 20: Volume Control on page 38</a> for range values.

The volume value is defined by the following formula:

Vol (dB) = Decimal value of LS\_C\_VOLUME\_MSB x 0.5 + Decimal value of LS\_C\_VOLUME\_LSB x 0.125 - 116 dB  
(each step is 0.125 dB).

**LS\_SUB\_VOLUME\_MSB**

Address: DEh

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

LS_SUB_VOLUME_MSB[7:0]
------------------------

Bit Name	Reset	Function
LS_SUB_VOLUME_MSB[7:0]	1001 1000	LS 10 bits volume Subwoofer channel 8 MSB See <a href="#">Figure 20: Volume Control on page 38</a> for range values.

**LS\_SUB\_VOLUME\_LSB**

Address: DFh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	0	LS_SUB_VOLUME_LSB[1:0]	

Bit Name	Reset	Function
Bits[7:2]	000000	Reserved.
LS_SUB_VOLUME_LSB[1:0]	00	LS 10 bits volume Subwoofer channel 2 LSB See <a href="#">Figure 20: Volume Control on page 38</a> for range values.

The volume value is defined by the following formula:

Vol (dB) = Decimal value of LS\_SUB\_VOLUME\_MSB x 0.5 + Decimal value of LS\_SUB\_VOLUME\_LSB x 0.125 - 116 dB (each step is 0.125 dB).

**LS\_SL\_VOLUME\_MSB**

Address: E0h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
LS_SL_VOLUME_MSB[7:0]							

Bit Name	Reset	Function
LS_SL_VOLUME_MSB[7:0]	1001 1000	LS 10 bits volume Left surround channel 8 MSB in independent mode or LS 10 bits Left and Right surround volume 8 MSB in differential mode. See <a href="#">Figure 20: Volume Control on page 38</a> or <a href="#">Figure 21: Differential Balance on page 39</a> .

**LS\_SL\_VOLUME\_LSB**

Address: E1h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	0	LS_SL_VOLUME_LSB[1:0]	

Bit Name	Reset	Function
Bits[7:2]	000000	Reserved.
LS_SL_VOLUME_LSB[1:0]	00	LS 10 bits volume Left surround channel 2 LSB in independent mode or LS 10 bits Left and Right surround volume 2 LSB in differential mode. See <a href="#">Figure 20: Volume Control on page 38</a> or <a href="#">Figure 21: Differential Balance on page 39</a> .

The volume value is defined by the following formula:

$$\text{Vol (dB)} = \text{Decimal value of LS_SL_VOLUME_MSB} \times 0.5 + \text{Decimal value of LS_SL_VOLUME_LSB} \times 0.125 - 116 \text{ dB}$$

(each step is 0.125 dB).

**LS\_SR\_VOLUME\_MSB**

Address: E2h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
LS_SR_VOLUME_MSB[7:0]							

Bit Name	Reset	Function
LS_SR_VOLUME_MSB[7:0]	00000000	LS 10 bits volume Right channel 8 MSB in independent mode or LS 10 bits surround Left and Right balance 8 MSB in differential mode. See <a href="#">Figure 20: Volume Control on page 38</a> or <a href="#">Figure 21: Differential Balance on page 39</a> .

**LS\_SR\_VOLUME\_LSB**

Address: E3h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	0	LS_SR_VOLUME_LSB[1:0]	

Bit Name	Reset	Function
Bits[7:2]	000000	Reserved.

Bit Name	Reset	Function
LS_SR_VOLUME_LSB[1:0]	00	LS 10 bits volume Right channel 8 MSB in independent mode or LS 10 bits surround Left and Right balance 2 LSB in differential mode. See <a href="#">Figure 20: Volume Control on page 38</a> or <a href="#">Figure 21: Differential Balance on page 39</a> .

The volume value is defined by the following formula:

Vol (dB) = Decimal value of LS\_SR\_VOLUME\_MSB x 0.5 + Decimal value of LS\_SR\_VOLUME\_LSB x 0.125 - 116 dB (each step is 0.125 dB).

### LS\_MASTER\_VOLUME\_MSB

Address: E4h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
LS_MASTER_VOLUME_MSB[7:0]							

Bit Name	Reset	Function
LS_MASTER_VOLUME_MSB[7:0]	1110100 0	LS 10 bits volume Master channel 8 MSB See <a href="#">Figure 20: Volume Control on page 38</a> for range values.

### LS\_MASTER\_VOLUME\_LSB

Address: E5h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	0	LS_MASTER_VOLUME_LSB[1:0]	

Bit Name	Reset	Function
Bits[7:2]	000000	Reserved.
LS_MASTER_VOLUME_LSB[1:0]	00	LS 10 bits volume Master channel 2 LSB See <a href="#">Figure 20: Volume Control on page 38</a> for range values.

The volume value is defined by the following formula:

Vol (dB) = Decimal value of LS\_MASTER\_VOLUME\_MSB x 0.5 + Decimal value of LS\_MASTER\_VOLUME\_LSB x 0.125 - 116 dB (each step is 0.125 dB).



**HP\_L\_VOLUME\_MSB**

Address: E6h

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

HP_L_VOLUME_MSB[7:0]
----------------------

Bit Name	Reset	Function
HP_L_VOLUME_MSB[7:0]	1001 1000	HP 10 bits volume Left channel 8 MSB in independent mode or HP 10 bits Left and Right volume 8 MSB in differential mode. See <a href="#">Figure 20: Volume Control on page 38</a> or <a href="#">Figure 21: Differential Balance on page 39</a> .

**HP\_L\_VOLUME\_LSB**

Address: E7h

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

0	0	0	0	0	0	HP_L_VOLUME_LSB[1:0]
---	---	---	---	---	---	----------------------

Bit Name	Reset	Function
Bits[7:2]	000000	Reserved.
HP_L_VOLUME_LSB[1:0]	00	HP 10 bits volume Left channel 2 LSB in independent mode or HP 10 bits Left and Right volume 2 LSB in differential mode. See <a href="#">Figure 20: Volume Control on page 38</a> or <a href="#">Figure 21: Differential Balance on page 39</a> .

The volume value is defined by the following formula:

$\text{Vol (dB)} = \text{Decimal value of HP\_L\_VOLUME\_MSB} \times 0.5 + \text{Decimal value of HP\_L\_VOLUME\_LSB} \times 0.125 - 116 \text{ dB}$   
(each step is 0.125 dB).

**HP\_R\_VOLUME\_MSB**

Address: E8h

Type: R/W

Bit 7      Bit 6      Bit 5      Bit 4      Bit 3      Bit 2      Bit 1      Bit 0

HP_R_VOLUME_MSB[7:0]
----------------------

Bit Name	Reset	Function
HP_R_VOLUME_MSB[7:0]	0000000 0	HP 10 bits volume Right channel 8 MSB in independent mode or HP 10 bits Left and Right balance 8 MSB in differential mode. See <a href="#">Figure 20: Volume Control on page 38</a> or <a href="#">Figure 21: Differential Balance on page 39</a> .

**HP\_R\_VOLUME\_LSB**

Address: E9h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	0	HP_R_VOLUME_LSB[1:0]	

Bit Name	Reset	Function
Bits[7:2]	000000	Reserved.
HP_R_VOLUME_LSB[1:0]	00	HP 10 bits volume Right channel 2 LSB in independent mode or HP 10 bits Left and Right balance 2LSB in differential mode. See <a href="#">Figure 20: Volume Control on page 38</a> or <a href="#">Figure 21: Differential Balance on page 39</a> .

**SCART\_L\_VOLUME\_MSB**

Address: EAh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
SCART_L_VOLUME_MSB[7:0]							

Bit Name	Reset	Function
SCART_L_VOLUME_MSB[7:0]	1101110 1	SCART 10 bits volume Left channel 8 MSB in independent mode or SCART10 bits Left and Right volume 8 MSB in differential mode. See <a href="#">Figure 20: Volume Control on page 38</a> or <a href="#">Figure 21: Differential Balance on page 39</a> .

**SCART\_L\_VOLUME\_LSB**

Address: EBh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	0	SCART_L_VOLUME_LSB[1:0]	

Bit Name	Reset	Function
Bits[7:2]	000000	Reserved.
SCART_L_VOLUME_LSB[1:0]	00	SCART 10 bits volume Left channel 2 LSB in independent mode or SCART10 bits Left and Right volume 2 LSB in differential mode. See <a href="#">Figure 20: Volume Control on page 38</a> or <a href="#">Figure 21: Differential Balance on page 39</a> .

The volume value is defined by the following formula:

$$\text{Vol (dB)} = \text{Decimal value of SCART\_L\_VOLUME\_MSB} \times 0.5 + \text{Decimal value of SCART\_L\_VOLUME\_LSB} \times 0.125 - 116 \text{ dB (each step is 0.125 dB)}$$
**SCART\_R\_VOLUME\_MSB**

Address: ECh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
SCART_R_VOLUME_MSB[7:0]							

Bit Name	Reset	Function
SCART_R_VOLUME_MSB[7:0]	11011101	SCART 10 bits volume Right channel 8 MSB in independent mode or SCART10 bits Left and Right balance 8 MSB in differential mode. See <a href="#">Figure 20: Volume Control on page 38</a> or <a href="#">Figure 21: Differential Balance on page 39</a> .

**SCART\_R\_VOLUME\_LSB**

Address: EDh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	0	SCART_R_VOLUME_LSB[1:0]	

Bit Name	Reset	Function
Bits[7:2]	000000	Reserved.

Bit Name	Reset	Function
SCART_R_VOLUME_LSB[1:0]	00	SCART 10 bits volume Right channel 2 LSB in independent mode or SCART10 bits Left and Right balance 2 LSB in differential mode. See <a href="#">Figure 20: Volume Control on page 38</a> or <a href="#">Figure 21: Differential Balance on page 39</a> .

## 12.19 Beeper

### BEEPER\_ON

### Beeper Activation Register

Address: EEh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	0	0	BEEPER_ON

Bit Name	Reset	Function
Bits [7:1]	0000000	Reserved.
BEEPER_ON	0	<b>Beeper Enable</b> 0: Beeper muted (Default.) 1: Beeper enabled.

### BEEPER\_MODE

### Beeper Control Register

Address: EFh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	BEEPER_DURATION	BEEPER_PULSE	BEEPER_PATH		

Bit Name	Reset	Function
Bits [7:5]	000	Reserved.
BEEPER_DURATION [4:3]	00	Define beeper duration when set to pulse mode.
BEEPER_PULSE	0	<b>Set beeper pulse mode</b> 0: Pulse mode selected. 1: Continuous mode selected.
BEEPER_PATH [1:0]	11	<b>Set the output channels when beeper is active</b> 00: no channels. 01: Loudspeakers only. 10: Headphone only. 11: Loudspeakers and Headphone selected.

**BEEPER\_FREQ\_VOL****Beeper Frequency and Volume Settings Register**

Address: F0h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
BEEP_FREQ[2:0]				BEEP_VOL[4:0]			

Bit Name	Reset	Function
BEEP_FREQ[2:0]	011	Defines the frequency of the beeper tone from 62.5 Hz to 8 kHz in octaves 000: 62.5 Hz                      100: 1 kHz 001: 125 Hz                      101: 2 kHz 010: 250 Hz                      110: 4 kHz 011: 500 Hz (Default)        111: 8 kHz
BEEP_VOL[4:0]	10000	Defines the Beeper volume from 0 to -93 dB in steps of 3 dB. 11111: 0 dB (1 V <sub>RMS</sub> )        ... 11110: -3 dB                      00011: -84 dB 11101: -6 dB                      00010: -87 dB ...                                    00001: -90 dB 10000: -48 dB (Default)       00000: -93 dB

**12.20 Mute****MUTE\_DIGITAL**

Address: F1h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
AUTOSTD_MUTE_ON	0	0	SCART_D_MUTE	SRND_HP_D_MUTE	SUB_D_MUTE	C_D_MUTE	LS_D_MUTE

Bit Name	Reset	Function
AUTOSTD_MUTE_ON	1	0: autostandard can not mute outputs 1: autostandard can mute outputs when no signal is detected
Bit s[6:5]	00	
SCART_D_MUTE	1	SCART left/right digital soft mute 0: signal un-muted 1: signal muted
SRND_HP_D_MUTE	1	LS Surround/HP left/right digital soft mute 0: signal un-muted 1: signal muted
SUB_D_MUTE	1	LS Subwoofer digital soft mute 0: signal un-muted 1: signal muted

Bit Name	Reset	Function
C_D_MUTE	1	LS Center digital soft mute 0: signal un-muted 1: signal muted
LS_D_MUTE	1	LS left/right digital soft mute 0: signal un-muted 1: signal muted

## 12.21 S/PDIF

### S/PDIF\_OUT\_CONFIG

### S/PDIF Output Configuration Register

Address: F2h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	S/PDIF_OUT_MUTE	S/PDIF_OUT_SELECT	

Bit Name	Reset	Function
Bits [7:3]	00000	Reserved.
S/PDIF_OUT_MUTE	1	S/PDIF Output Mute: 0: S/PDIF Output unmuted. 1: S/PDIF Output muted.
S/PDIF_OUT_SELECT[1:0]	00	S/PDIF Output channel selection: 00: output SCART signal 01: output LS L-R signal 10: output C/SUB signal 11: output Sur/HP signal

## 12.22 Headphone Configuration

### HEADPHONE\_CONFIG

### Headphone Configuration Register

Address: F3h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	HP_FORCE	HP_LS_MUTE	HP_DET_ACTIVE	HP_DETECTED

Bit Name	Reset	Function
Bits [7:4]	0000	Reserved.

Bit Name	Reset	Function
HP_FORCE	0	1: force output of the HP signal (bypass surround)
HP_LS_MUTE	0	0: when HP is detected and active, LS are not muted 1: when HP is detected and active, LS are muted
HP_DET_ACTIVE	1	0: HP detection is not active 1: HP detection is active, when HP detected, Surround signal is bypassed and HP signal is output on HP
HP_DETECTED	0	1: When a signal is detected on HP_DET pin (STATUS)

## 12.23 DAC Control

### DAC\_CONTROL

### DAC Control Register

Address: F4h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	S/PDIF_MUX	DAC_SCART_MUTE	DAC_SHP_MUTE	DAC_CSUB_MUTE	DAC_LSLR_MUTE	POWER_UP

Bit Name	Reset	Function
Bits [7:6]	00	Reserved.
S/PDIF_MUX	0	redirect external or internal S/PDIF source to S/PDIF output : 0: internal S/PDIF 1: external S/PDIF
DAC_SCART_MUTE	1	SCART left/right analog soft mute 0: signal un-muted 1: signal muted
DAC_SHP_MUTE	1	Surround/HP left/right analog soft mute 0: signal un-muted 1: signal muted
DAC_CSUB_MUTE	1	Center/Subwoofer analog soft mute 0: signal un-muted 1: signal muted
DAC_LSLR_MUTE	1	LS left/right analog soft mute 0: signal un-muted 1: signal muted
POWER_UP	1	0: DACs Power OFF 1: Power ON

**DAC\_SW\_CHANNELS****DAC Switch Channels Register**

Address: F5h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
SUR_HP_SW	C_SUB_SW		LS_L_R_SW		SCART_SW		

Bit Name	Reset	Function
SUR_HP_SW	00	HP/Surround DAC: 00: Left/Right channels non inverted 11: Left/Right channels inverted
C_SUB_SW	00	Center/SubDAC: 00: Left/Right channels non inverted 11: Left/Right channels inverted
LS_L_R_SW	00	LS Left-Right DAC: 00: Left/Right channels non inverted 11: Left/Right channels inverted
SCART_SW	00	SCART DAC: 00: Left/Right channels non inverted 11: Left/Right channels inverted

**SPDIF\_SW\_CHANNELS****SPDIF Switch Channels Register**

Address: F6h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	0	SPDIF_SW	

Bit Name	Reset	Function
Bits [7:2]	000000	Reserved.
SPDIF_SW	00	SPDIF output: 00: Left/Right channels non inverted 11: Left/Right channels inverted

**SPDIF\_CHANNEL\_STATUS**

Address: F9h

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
CHANNEL_STATUS		EMPHASIS			COPYRIGHT	NON_AUDIO	PRO_CON



Bit Name	Reset	Function
CHANNEL_STATUS[7:6]	00	Channel status mode: 00: Mode zero other values: reserved
EMPHASIS[5:3]	000	Emphasis: according to IEC60958 specification
COPYRIGHT	0	Copyright: 0: Asserted 1: Not asserted
NON_AUDIO	0	Non-audio: 0: Linear PCM 1: Non-audio signal
PRO_CON	0	Select Professional or Consumer modes: 0: Consumer 1: Professional

## 12.24 AutoStandard Coefficients Settings

### AUTOSTD\_COEFF\_CTRL

Address: FBh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	0	AUTOSTD_COEFF_CTRL[1:0]	

Bit Name	Reset	Function
Bits [7:2]	000000	Reserved.
AUTOSTD_COEFF_CTRL[1:0]	01	Control the Demod filter coeff table settings 00: No action 01: Init Coefficients to ROM values 10: Update Coefficients with I2C values (set to 0 by DSP to acknowledge)

### AUTOSTD\_COEFF\_INDEX\_MSB

Address: FCh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	0	0	0	0	0	0	AUTOSTD_COEFF_INDEX_MSB

Bit Name	Reset	Function
Bits [7:2]	0000000	Reserved.
AUTOSTD_COEFF_INDEX_MSB	0	FIR Coefficients table index (MSB)

### AUTOSTD\_COEFF\_INDEX\_LSB

Address: FDh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
AUTOSTD_COEFF_INDEX_LSB[7:0]							

Bit Name	Reset	Function
AUTOSTD_COEFF_INDEX_LSB[7:0]	0000 0000	FIR Coefficients table index (LSB)

### AUTOSTD\_COEFF\_VALUE

Address: FEh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
AUTOSTD_COEFF_VALUE[7:0]							

Bit Name	Reset	Function
AUTOSTD_COEFF_VALUE[7:0]	0000 0000	Reserved

**Note:** These four registers (AUTOSTD\_COEFF\_CTRL, AUTOSTD\_COEFF\_INDEX\_MSB, AUTOSTD\_COEFF\_INDEX\_LSB and AUTOSTD\_COEFF\_VALUE) can be used to change parameter settings for the following parts of channel 1 or channel 2:

- Channel carrier DCO frequency (register CARFQxx)
- Channel filter coefficients (registers FIRxCx)
- PLL baseband AM/FM demodulators proportional and integral coefficients (registers ACOEFFx or BCOEFFx)
- Demodulator mode selection (register DEMOD\_CTRL)
- IF AGC control (AGC\_CTRL)
- Channel 2 symbol tracking loop parameters (register SCOEFF)
- Zweiton control (register ZWT\_CTRL)

**While keeping the AUTOSTANDARD function always active.**  
New values for all parameters mentioned above are kept instead of the values automatically sent by the AUTOSTANDARD function.

One application is for example to implement OVERMODULATION recovery mode for any sound standard supported by the device (B/G, I, M/N, DK1, DK2, or DK3).  
 See *Technical Note* for instructions on how to update the coefficient table settings.

## PATCH\_VERSION

Address: FFh

Type: R/W

Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
PATCH_VERSION[7:0]							

Bit Name	Reset	Function
PATCH_VERSION[7:0]	0000 0000	Indicate the patch version which has been loaded in the device (can be used to check if the patch has been correctly loaded)

## 13 Electrical Characteristics

Test Conditions:  $T_{OPER} = 25^{\circ}C$ ,  $V_{CC\_H} = 8V$ ,  $V_{XX\_18} = 1.8V$ ,  $V_{XX\_33} = 3.3V$ , Oscillator at 27 MHz, default register values for synthesizer, otherwise specified.

### 13.1 Absolute Maximum Ratings

Symbol	Parameter	Value	Units
$V_{XX\_18}$	Analog and Digital 1.8 V Supply Voltage ( $V_{CC18\_CLK1}$ , $V_{CC18\_CLK2}$ , $V_{CC18\_IF}$ , $V_{DD18}$ , $V_{DD18\_CONV}$ , $V_{DD18\_ADC}$ )	2.5	V
$V_{XX\_33}$	Analog and Digital 3.3 V Supply Voltage ( $V_{CC33\_SC}$ , $V_{CC33\_LS}$ , $V_{DD33\_IO1}$ , $V_{DD33\_IO2}$ , $V_{DD33\_CONV}$ , $V_{CC\_NISO}$ )	4.0	V
$HV_{CC}$	Analog Supply High Voltage ( $V_{CC\_H}$ )	8.8	V
$V_{ESD}$	Capacitor 100 pF discharged via 1.5 k $\Omega$ serial resistor (Human Body Model)	4	kV
$T_{OPER}$	Operating Ambient Temperature	0, +70	$^{\circ}C$
$T_{STG}$	Storage Temperature	-55 to +150	$^{\circ}C$

### 13.2 Thermal Data

Symbol	Parameter	Value	Units
$R_{thJA}$	Junction-to-Ambient Thermal Resistance	42	$^{\circ}C/W$

### 13.3 Power Supply Data

Symbol	Parameter	Min.	Typ.	Max.	Units
$V_{XX\_18}$	Analog and Digital 1.8 V Supply Voltage ( $V_{CC18\_CLK1}$ , $V_{CC18\_CLK2}$ , $V_{CC18\_IF}$ , $V_{DD18}$ , $V_{DD18\_CONV}$ , $V_{DD18\_ADC}$ )	1.70	1.80	1.90	V
$V_{XX\_33}$	Analog and Digital 3.3 V Supply Voltage ( $V_{CC33\_SC}$ , $V_{CC33\_LS}$ , $V_{DD33\_IO1}$ , $V_{DD33\_IO2}$ , $V_{DD33\_CONV}$ , $V_{CC\_NISO}$ )	3.13	3.30	3.47	V
$HV_{CC}$	Analog Supply High Voltage ( $V_{CC\_H}$ )	7.6	8.0	8.4	V
$I_{VDD18}$	Current Consumption for Digital 1.8 V Supply ( $V_{CC18\_CLK2}$ , $V_{DD18}$ , $V_{DD18\_CONV}$ , $V_{DD18\_ADC}$ )		230	280	mA
$I_{VDD33}$	Current Consumption for Digital 3.3 V Supply ( $V_{DD33\_IO1}$ , $V_{DD33\_IO2}$ )		10	12	mA
$I_{VCC18}$	Current Consumption for Analog 1.8 V Supply ( $V_{CC18\_CLK1}$ , $V_{CC18\_IF}$ )		50	60	mA
$I_{VCC33}$	Current Consumption for Analog 3.3 V Supply ( $V_{CC33\_SC}$ , $V_{CC33\_LS}$ , $V_{DD33\_CONV}$ , $V_{CC\_NISO}$ )		65	78	mA
$I_{VCC\_H}$	Current Consumption for Analog Supply High Voltage (8 V)		4	7	mA
$P_{DTOT}$	Total Power Dissipation		780	965	mW

### 13.4 Crystal Oscillator

Symbol	Parameter	Min.	Typ.	Max.	Units
$f_P$	Crystal Series Resonance Frequency (at $C_{21} = C_{22} = 27$ pF load capacitor)		27		MHz
$DF/F_P$	Frequency Tolerance at 25 °C	-30		+30	ppm
$DF/F_T$	Frequency Stability versus Temperature within a range from 0 to 70 °C	-30		+30	ppm
$C_1$	Motional Capacitor			15	fF
$R_S$	Serial Resistance			30	$\Omega$
$C_S$	Shunt Capacitance			7	pF

### 13.5 Analog Sound IF Signal

Symbol	Parameter	Test Conditions	Min.	Typ.	Max.	Units
$BAND_{SIF}$	SIF Frequency Flatness	AGC_ERR at 0, frequency range from 4 to 7 MHz		0.6	3	dB
$R_{INSIF}$	SIF Input Resistance		60	72	85	k $\Omega$
$DC_{INSIF}$	SIF Input DC Level			0.9		V
$C_{INSIF}$	SIF Input Capacitance			3		pF
<b>FM Carrier</b>						
$VSIF_{FM}$	SIF Input Sensitivity	SNR 40dB RMS unweighted, 20 Hz-15 kHz, Standard B/G 27 kHz FM Deviation, 1 kHz	350			$\mu V_{PP}$
$DEV_{FM}$	FM Maximum Deviation	FM50k (Standard)	$\pm 15$	$\pm 50$	$\pm 115$	kHz
		FM200k		$\pm 200$	$\pm 320$	
		FM350k		$\pm 350$	$\pm 560$	
		FM500k		$\pm 500$	$\pm 700$	
$DFSIF_{FM}$	SIF Carrier Accuracy for FM	Standard (FM50k)		$\pm 1$	$\pm 5$	kHz
		Shifted Standard (FM50k with DCO compensation)			$\pm 120$	kHz
$R_{FM/QPSK}$	Carrier Ratio FM/QPSK for NICAM System	NICAM mute, FAR_MODE is active, standard BG, 100 mV <sub>PP</sub> level for FM carrier			40	dB
<b>AM Carrier</b>						
$VSIF_{AM}$	SIF Input Sensitivity	Unmodulated, -3 dB at output amplitude AGC_ERR at 21d Standard L, 54% AM Depth, 1 kHz	19			mV <sub>PP</sub>
$V_{MAX\_SIF_{AM}}$	SIF Maximum Input Level	Unmodulated, THD at 1%, 54% AM Depth, AGC_ERR at 0			1.3	V <sub>PP</sub>
$DEV_{AM}$	Modulation Depth for AM	THD at 1%	0		100	%

Symbol	Parameter	Test Conditions	Min.	Typ.	Max.	Units
DFSIF <sub>AM</sub>	SIF Carrier Accuracy for AM			±1	±5	kHz
R <sub>AM/QPSK</sub>	AM/QPSK Carrier Ratio for NICAM System	NICAM Mute, 100 mV <sub>PP</sub> AM carrier			36	dB
<b>AGC</b>						
AGC <sub>step</sub>	IF AGC Step		1.4	1.5	1.6	dB
AGC <sub>dyn</sub>	Relative maximum gain to step 0	Valid from step 21 to step 31	29	30	31	dB

### 13.6 SIF to I<sup>2</sup>S Output Path Characteristics

Test Conditions: SIF amplitude = 10 mVpp, otherwise specified, I<sup>2</sup>S output.

Symbol	Parameter	Test Conditions	Min.	Typ.	Max.	Units
<b>FM Demodulation</b>						
BAND <sub>FM</sub>	Frequency Response	20 Hz - 15 kHz			±0.7	dB
SNR <sub>FM</sub>	Signal to Noise	RMS unweighted, 20 Hz-15k Hz, Standard B/G 27 kHz FM Deviation, 1 kHz	66			dB
THD <sub>FM</sub>	Total Harmonic Distortion				0.05	%
SEP <sub>FM</sub>	Stereo Channel Separation	Standard B/G stereo A2, 27 kHz FM deviation, 1 kHz	48			dB
<b>NICAM Demodulation</b>						
BAND <sub>NIC</sub>	Frequency Response	20 Hz - 15 kHz			±0.2	dB
SNR <sub>NIC</sub>	Signal to Noise	200 Hz - 60 dBFS, trap filter 200 Hz RMS unweighted, 20 Hz-15 kHz, Standard B/G mono NICAM, 1 kHz	74			dB
THD <sub>NIC</sub>	Total Harmonic Distortion				0.04	%
<b>AM Demodulation</b>						
BAND <sub>AM</sub>	Frequency Response	20 Hz - 15 kHz			±0.5	dB
SNR <sub>AM</sub>	Signal to Noise	RMS unweighted 20 Hz-15 kHz, Standard L, 54% AM Depth, 1 kHz AGC: 13d	60			dB
THD <sub>AM</sub>	Total Harmonic Distortion				0.4	%

### 13.7 SCART to SCART Analog Path Characteristics

Test Conditions: R<sub>load</sub><sub>MAX</sub> = 10kΩ, C<sub>load</sub><sub>MAX</sub> = 330pF, MONO\_IN voltage = 0.5 V<sub>RMS</sub>

Symbol	Parameter	Test Conditions	Min.	Typ.	Max.	Units
<b>Analog-to-Analog STEREO and MONO</b>						
R <sub>INSCART</sub>	SCART Input Resistance		29	34	39	kΩ
R <sub>OUTSCART</sub>	Output Resistance for SCARTs			40	75	Ω
VDC <sub>INSCART</sub>	SCART Input DC Level		1.45	1.57	1.65	V
VDC <sub>OUTSCART</sub>	SCART Output DC Level		3.4	3.64	3.8	V

Symbol	Parameter		Test Conditions	Min.	Typ.	Max.	Units
CLIP <sub>SCART</sub>	Clipping SCART	Clipping input level from SCART input	At 1 kHz 1% THD	2.0			V <sub>RMS</sub>
		Clipping input level from MONO_IN input		0.5			V <sub>RMS</sub>
THD <sub>SCART</sub>	THD SCART	THD from SCART input	1 V <sub>RMS</sub> , at 1 kHz		0.02	0.05	%
		THD from MONO_IN input	0.25 V <sub>RMS</sub> , at 1 kHz		0.02	0.05	%
SNR <sub>SCART</sub>	Signal to Noise Ratio	SCART input	1 V <sub>RMS</sub> , 20 Hz to 20 kHz Bandwidth, RMS unweighted	82	90		dB
		MONO_IN input	0.25 V <sub>RMS</sub> , 20 Hz to 20 kHz Bandwidth, RMS unweighted	82	90		dB
BAND <sub>SCART</sub>	Frequency Flatness	SCART input	20 Hz to 20 kHz	-0.5	0	0.5	dB
		MONO_IN input	20 Hz to 20 kHz	11.5	12	12.5	dB
XTALK <sub>L/R</sub>	Left/Right Crosstalk		1 V <sub>RMS</sub> @ 1 kHz on ref signal, the other one grounded	80	90		dB
XTALK <sub>IN</sub>	Audio Crosstalk from Input Channel <i>n</i> to Input Channel <i>m</i>		1 V <sub>RMS</sub> @ 1 kHz on ref signal, all other inputs grounded	80	90		dB
XTALK <sub>OUT</sub>	Audio Crosstalk from Output Channel <i>n</i> to Output Channel <i>m</i>		1 V <sub>RMS</sub> @ 1 kHz on reference output, signal on a single input, all other inputs grounded	80	90		dB

### 13.8 SCART and MONO IN to I<sup>2</sup>S Path Characteristics

Test Conditions: Sampling Frequency = 32 kHz, Maximum MONO\_IN voltage = 0.5 V<sub>RMS</sub>.

Symbol	Parameter		Test Conditions	Min.	Typ.	Max.	Units
THD <sub>ADC</sub>	THD ADC	THD from SCART input	V <sub>IN</sub> = 2 V <sub>RMS</sub> at 1 kHz		0.006	0.05	%
		THD from MONO_IN input	V <sub>IN</sub> = 0.5 V <sub>RMS</sub> at 1 kHz		0.006	0.05	%
SNR <sub>ADC</sub>	Signal to Noise Ratio		20 to 15 kHz Bandwidth, RMS unweighted V <sub>IN</sub> = 200 mV <sub>RMS</sub> SCART input	62			dB
BAND <sub>ADC</sub>	Frequency Flatness		20 Hz to 15 kHz			±0.5	dB
XTALK <sub>ADC</sub>	Left Right Crosstalk		at 1 kHz, V <sub>IN</sub> = 1 V <sub>RMS</sub>	95			dB

### 13.9 I<sup>2</sup>S to LS/HP/SUB/C Path Characteristics

Test Conditions: Sampling Frequency = 32 kHz, L<sub>LOAD</sub> = 100 μH, C<sub>LOAD</sub> = 33 nF, R<sub>LOAD</sub> = 30 KΩ

Symbol	Parameter		Test Conditions	Min.	Typ.	Max.	Units
R <sub>OUTDAC</sub>	Output Resistance for Main Outputs		LS_L, LS_R, LS_SUB, LS_C, HP_LSS_R and HP_LSS_L pins		90	140	Ω
VDC <sub>OUTDAC</sub>	MAIN Output DC Level			1.4	1.55	1.8	V

Symbol	Parameter	Test Conditions	Min.	Typ.	Max.	Units
THD <sub>DAC</sub>	Total Harmonic Distortion	90% Full-scale Range at 1 kHz			0.06	%
SNR <sub>DAC</sub>	Signal to Noise Ratio	20 to 15 kHz Bandwidth, RMS unweighted, at -20 dB full range	75			dB
V <sub>OUTAMPDAC</sub>	MAIN Output Amplitude	100% Full-scale Range at 1 kHz	800	900	1050	mV <sub>RMS</sub>
XTALK <sub>DAC</sub>	Left Right Crosstalk	at 1 kHz, -20 dBFS	87			dB

### 13.10 I<sup>2</sup>S to SCART Path Characteristics

Test Conditions: Sampling Frequency = 32 kHz, C<sub>LOAD</sub> = 33nF on DAC SCART pins, DAC SCART prescale at -5.5 dB.

Symbol	Parameter	Test Conditions	Min.	Typ.	Max.	Units
THD <sub>DACSCART</sub>	Total Harmonic Distortion	90% Full-scale Range at 1 kHz		0.08	0.12	%
SNR <sub>DACSCART</sub>	Signal to Noise Ratio	20 Hz to 15 kHz Bandwidth unweighted, -20 dB Full Range	73			dB
V <sub>ODACSCART</sub>	MAIN Output Amplitude	100% Full-scale Range at 1 kHz	1.75	2	2.25	V <sub>RMS</sub>
XTALK <sub>DACSCART</sub>	Left Right Crosstalk	at 1 kHz, -20 dBFS	80			dB

### 13.11 MUTE Characteristics

Symbol	Parameter	Test Conditions	Min.	Typ.	Max.	Units
MUTE <sub>DAC</sub>	DAC Mute analog	I <sup>2</sup> S to DAC at 1 kHz	90			dB
MUTE <sub>SCART</sub>	SCART Mute	2 V <sub>RMS</sub> @ 1 kHz on ref signal, all other inputs grounded	81			dB

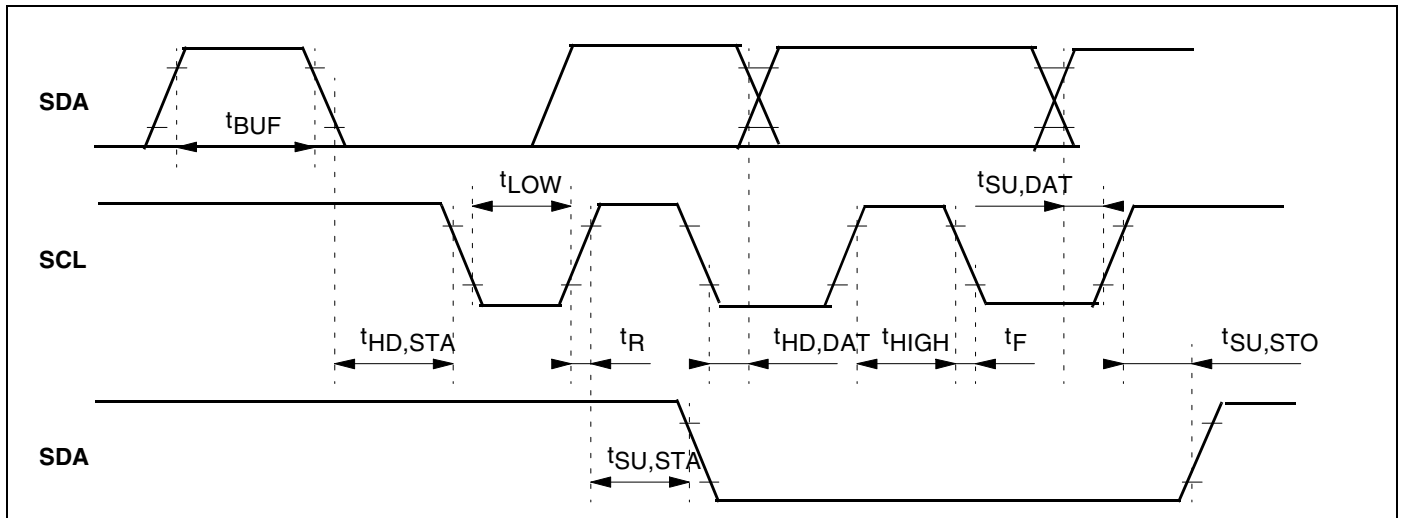
### 13.12 Digital I/Os Characteristics

Symbol	Parameter	Test Conditions	Min.	Typ.	Max.	Units
V <sub>IL</sub>	Low Level Input Voltage	Except SDA, SCL and CLK_SEL, 3.3 V power supply			0.5	V
V <sub>IH</sub>	High Level Input Voltage	Except SDA, SCL and CLK_SEL, 3.3 V power supply	2.0			V
I <sub>IN</sub>	Input Current				1	μA
V <sub>ILCLK_SEL</sub>	CLK_SEL Low Level Input Voltage	1.8 V power supply			0.3	V
V <sub>IHCLK_SEL</sub>	CLK_SEL High Level Input Voltage	1.8 V power supply	1.2			V
V <sub>OL</sub>	Low Level Output Voltage	S/PDIF_OUT, IRQ, BUS_EXP			0.3	V
V <sub>OH</sub>	High Level Output Voltage	S/PDIF_OUT, IRQ, BUS_EXP	3.0			V



13.13 I<sup>2</sup>C Bus Characteristics

Symbol	Parameter	Test Conditions	Min.	Typ	Max.	Unit
<b>SCL</b>						
V <sub>IL</sub>	Low Level Input Voltage		-0.3		1.5	V
V <sub>IH</sub>	High Level Input Voltage		2.3		5.5	V
I <sub>IL</sub>	Input Leakage Current	V <sub>IN</sub> = 0 to 5.0 V	-10		10	μA
f <sub>SCL</sub>	Clock Frequency				400	kHz
t <sub>R</sub>	Input Rise Time	1 V to 2 V			300	ns
t <sub>F</sub>	Input Fall Time	2 V to 1 V			300	ns
C <sub>I</sub>	Input Capacitance				10	pF
<b>SDA</b>						
V <sub>IL</sub>	Low Level Input Voltage		-0.3		1.5	V
V <sub>IH</sub>	High Level Input Voltage		2.3		5.5	V
I <sub>IL</sub>	Input Leakage Current	V <sub>IN</sub> = 0 to 5.0 V	-10		10	μA
t <sub>R</sub>	Input Rise Time	1 V to 2 V			300	ns
t <sub>F</sub>	Input Fall Time	2 V to 1 V			300	ns
V <sub>OL</sub>	Low Level Output Voltage	I <sub>OL</sub> = 3 mA			0.4	V
t <sub>F</sub>	Output Fall Time	2 V to 1 V			250	ns
C <sub>L</sub>	Load Capacitance				400	pF
C <sub>I</sub>	Input Capacitance				10	pF
<b>I<sup>2</sup>C Timing</b>						
t <sub>LOW</sub>	Clock Low period		1.3			μs
t <sub>HIGH</sub>	Clock High period		0.6			μs
t <sub>SU,DAT</sub>	Data Set-up Time		100			ns
t <sub>HD,DAT</sub>	Data Hold Time		0		900	ns
t <sub>SU,STO</sub>	Set-up Time from Clock High to Stop		0.6			μs
t <sub>BUF</sub>	Start Set-up Time following a Stop		1.3			μs
t <sub>HD,STA</sub>	Start Hold Time		0.6			μs
t <sub>SU,STA</sub>	Start Set-up Time following Clock Low to High Transition		0.6			μs

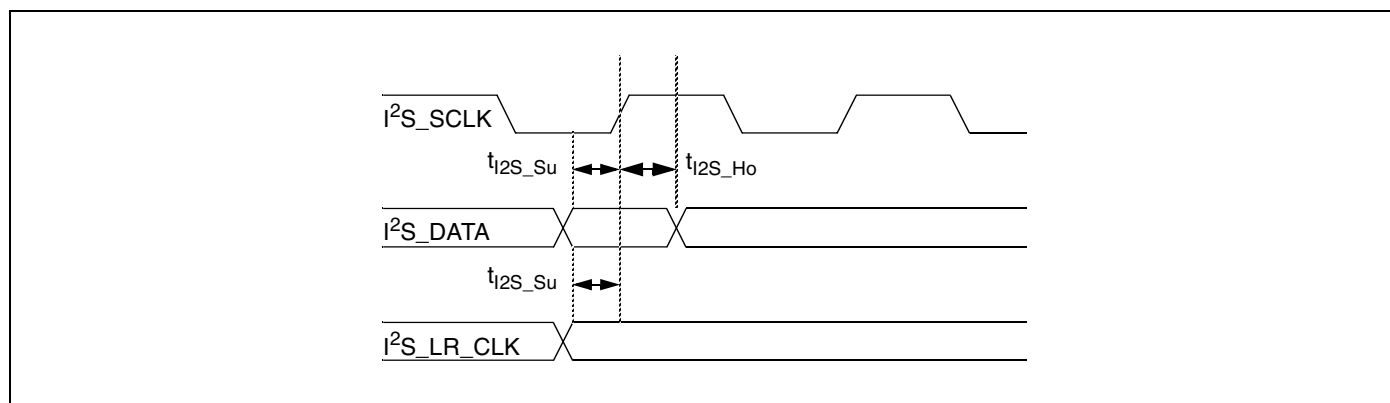
Figure 34: I<sup>2</sup>C Bus Timing

### 13.14 I<sup>2</sup>S Bus Interface

See timing for I<sup>2</sup>S on page 43.

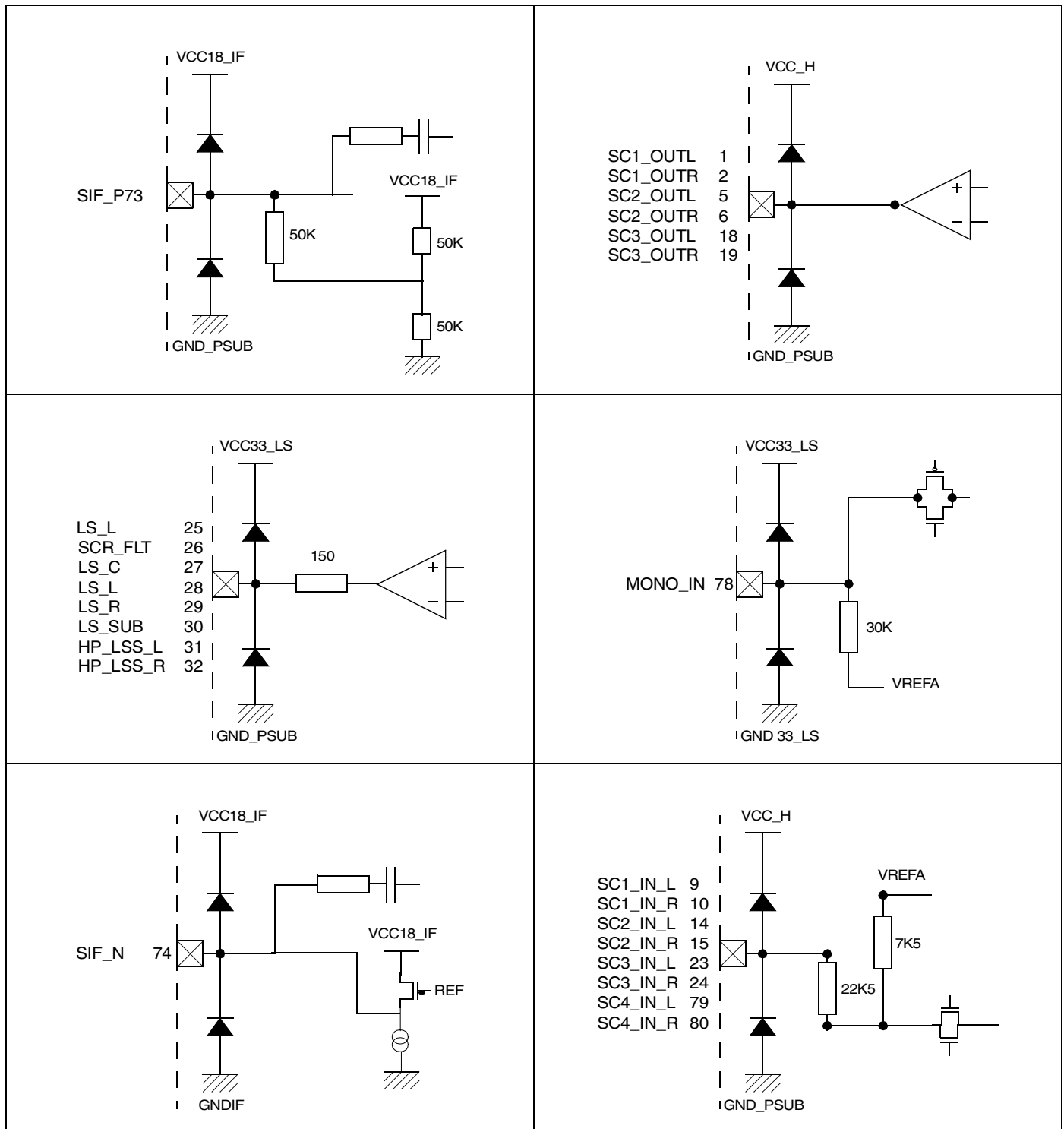
Symbol	Parameter	Test Conditions	Min.	Typ	Max.	Unit
<b>I<sup>2</sup>S Input</b>						
$V_{I2S\_IL}$	Input I <sup>2</sup> S Low Level Voltage				0.8	V
$V_{I2S\_IH}$	Input I <sup>2</sup> S High Level Voltage		2			V
$Z_{I2S}$	Input I <sup>2</sup> S Impedance				5	pF
$I_{I2S\_Leak}$	I <sup>2</sup> S Leakage Current		-1		1	μA
$t_{I2S\_Su}$	I <sup>2</sup> S Input Setup Time before Rising Edge of Clock	See Figure 35	30			ns
$t_{I2S\_Ho}$	I <sup>2</sup> S Input Hold Time after Rising Edge of Clock	See Figure 35	100			ns
$f_{I2S\_LR0}$	I <sup>2</sup> S Left Right Strobe Input Frequency (I <sup>2</sup> S_DATA0 only)	Deviation = ±250 ppm	8		48	kHz
$f_{I2S\_SCL0}$	I <sup>2</sup> S Serial Clock Input Frequency (I <sup>2</sup> S_DATA0 only)		0.512		3.072	MHz
$f_{I2S\_LR}$	I <sup>2</sup> S Left Right Strobe Input Frequency (I <sup>2</sup> S_DATA0,1,2)	Deviation = ±250 ppm	32		48	kHz
$f_{I2S\_SCL}$	I <sup>2</sup> S Serial Clock Input Frequency (I <sup>2</sup> S_DATA0,1,2)		2.048		3.072	MHz
$R_{I2S\_SCL}$	I <sup>2</sup> S Serial Clock Input Ratio		0.9		1.1	
<b>I<sup>2</sup>S Output (I<sup>2</sup>S_DATA0 only)</b>						
$V_{I2SOL}$	Output I <sup>2</sup> S Low Level Voltage	IOL = 2 mA			0.4	V
$V_{I2SOH}$	Output I <sup>2</sup> S High Level voltage	IOH = 2 mA	2.4			V
$f_{I2S\_OLR}$	I <sup>2</sup> S Left Right Strobe Output Frequency	Deviation = ±250 ppm	8		48	kHz

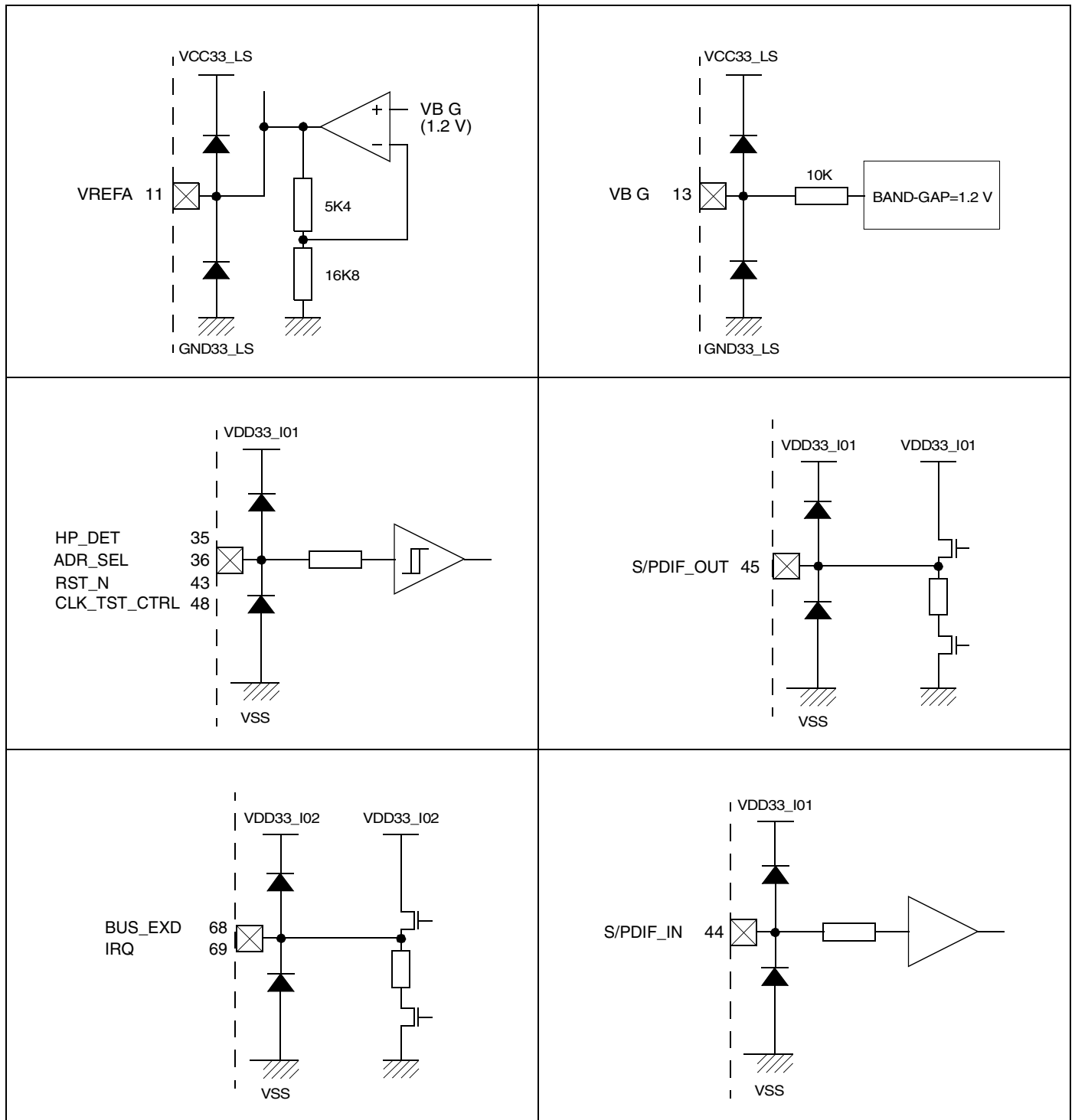
Symbol	Parameter	Test Conditions	Min.	Typ	Max.	Unit
$f_{I2S\_OSCI}$	I <sup>2</sup> S Serial Clock Output Frequency		0.512		3.072	MHz
$R_{I2S\_SCL}$	I <sup>2</sup> S Serial Clock Output Ratio		0.9		1.1	
$t_{I2S\_DeI}$	I <sup>2</sup> S Output Delay After Falling Edge of Clock	See Figure 35, $C_{LOAD} = 30$ pF			30	ns

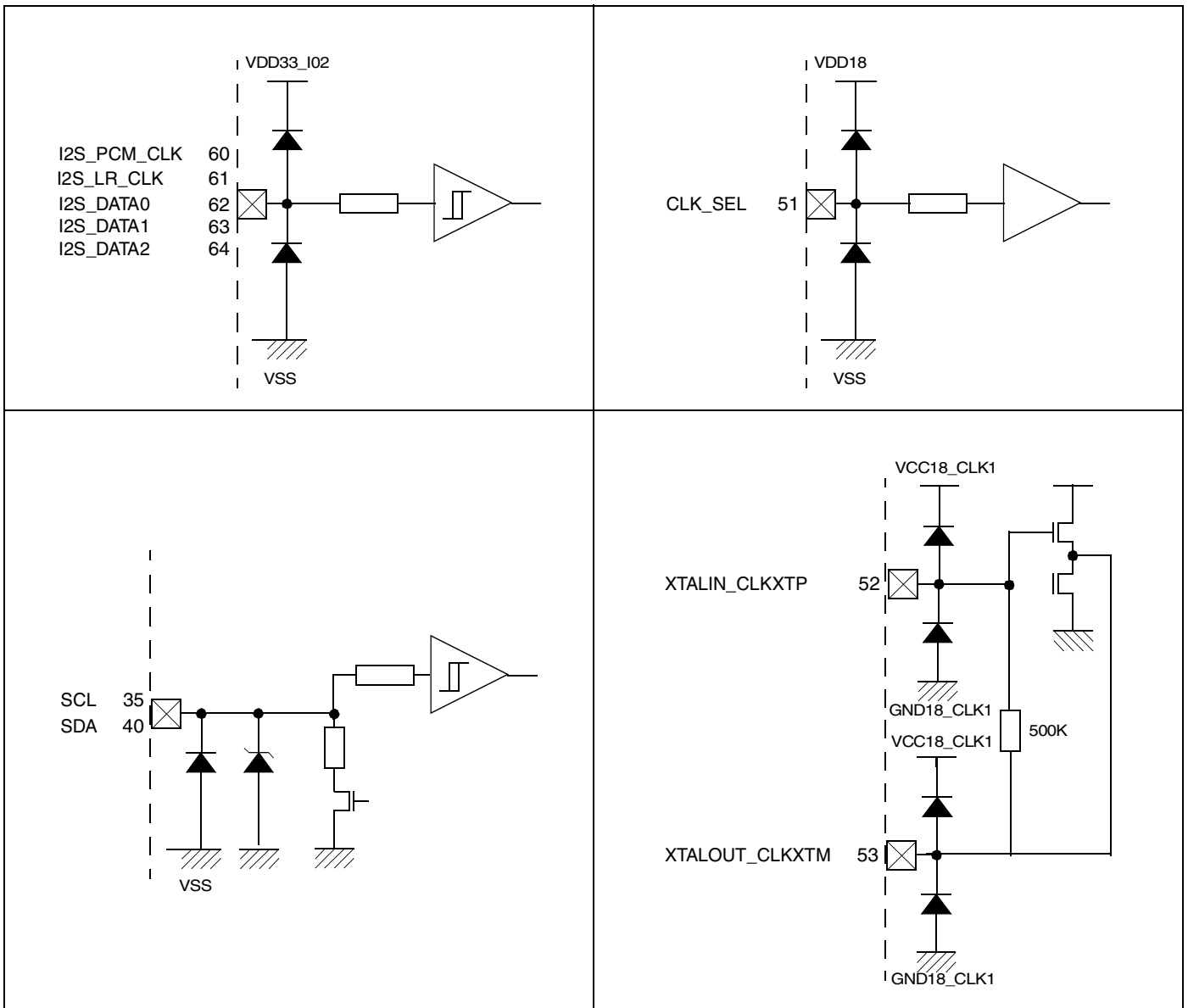
Figure 35: I<sup>2</sup>S Input Bus Timing

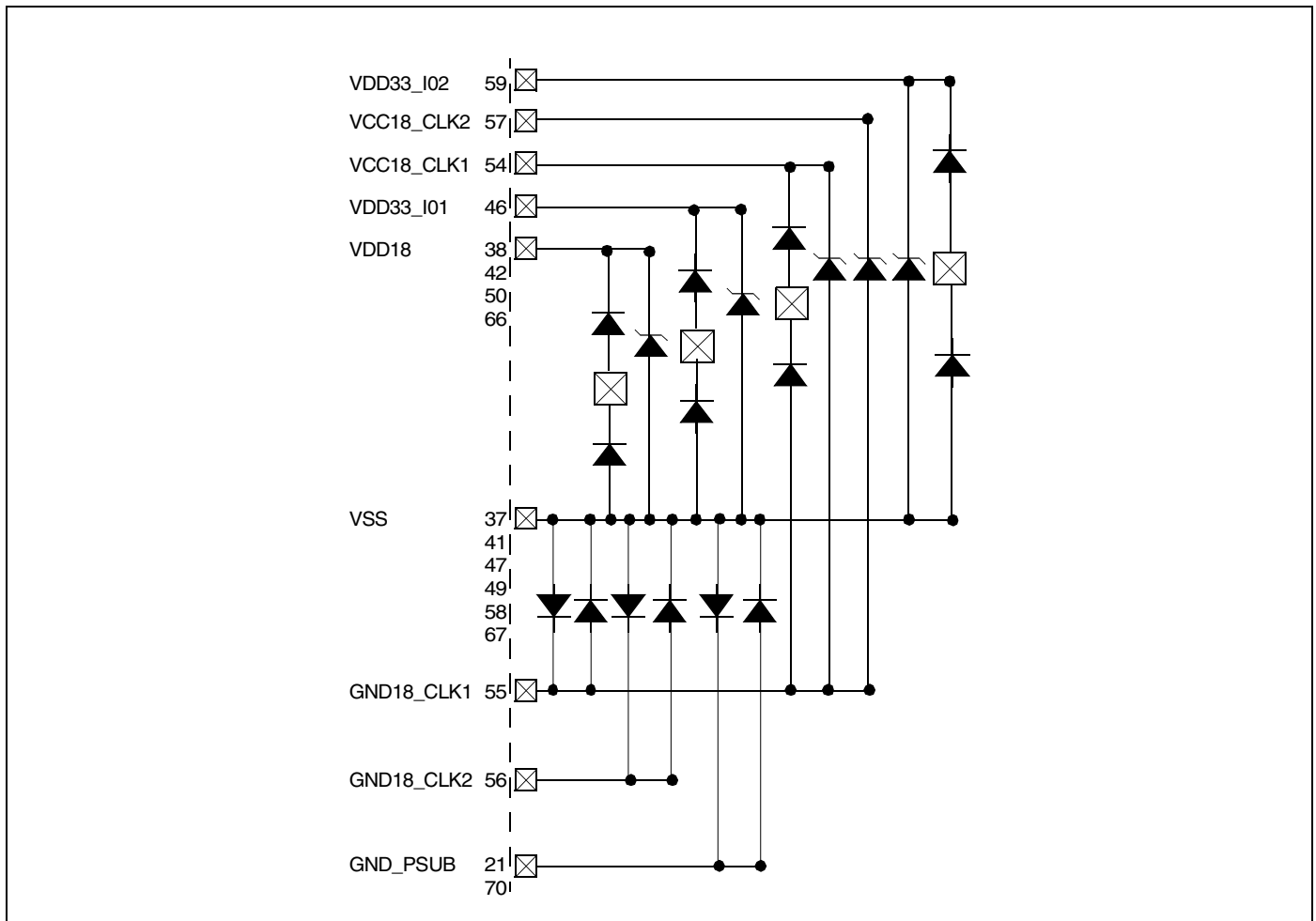
# 14 Input/Output Groups

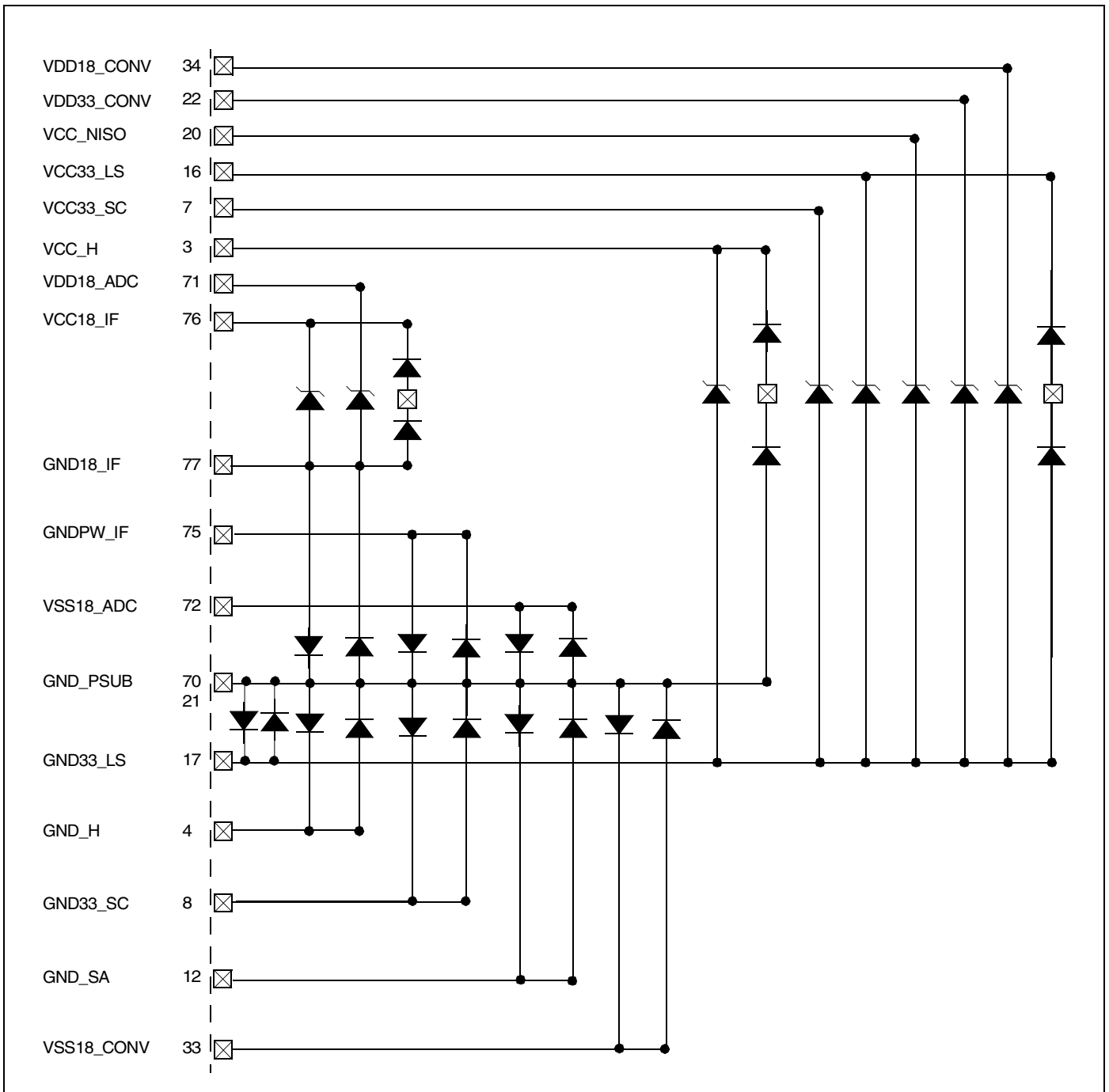
Pin numbers apply to SDIP package only.













## 15 Package Mechanical Data

Figure 36: 80-Pin Thin Plastic Quad Flat Package

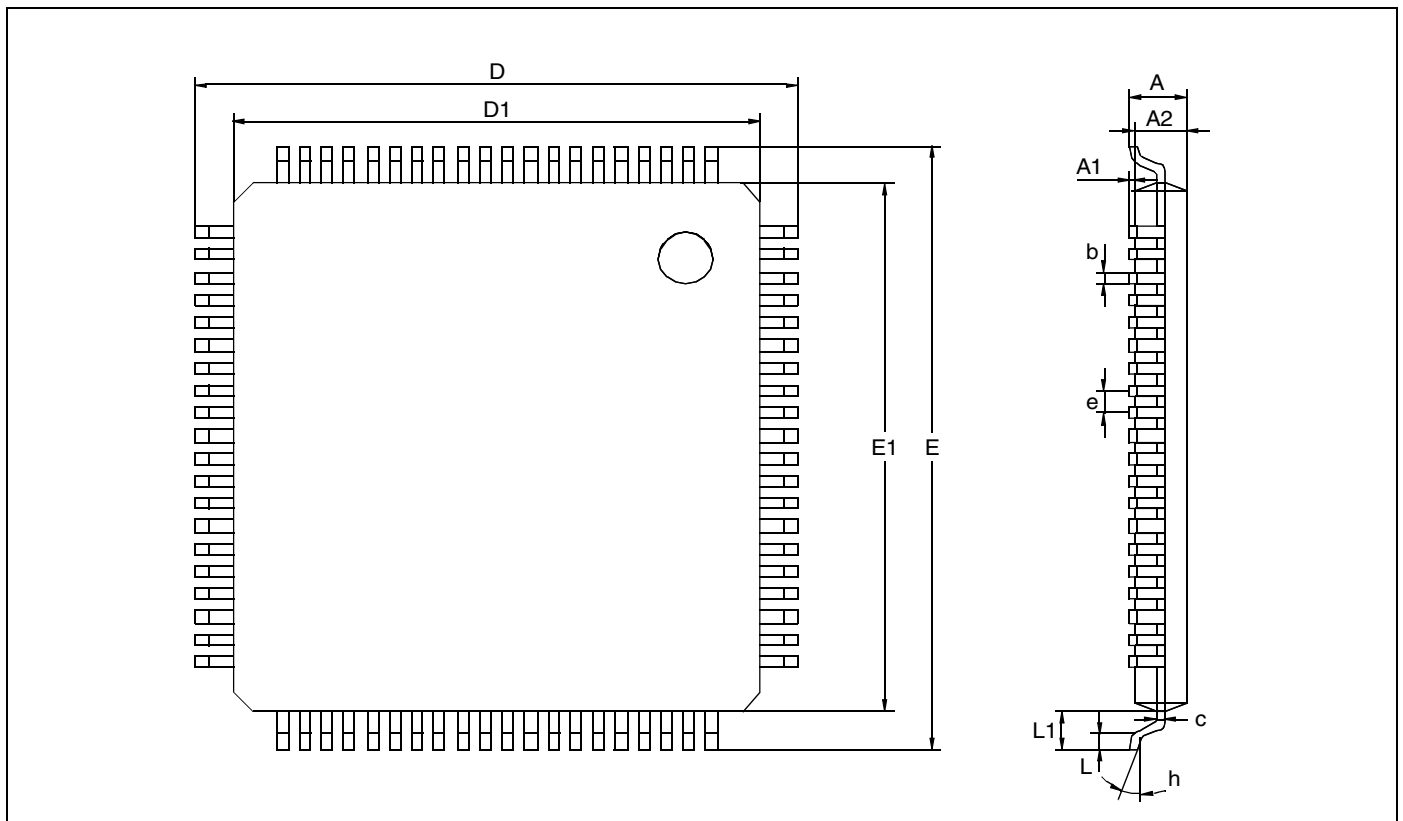


Table 32: Package Mechanical Dimensions

Dim.	mm			inches		
	Min.	Typ.	Max.	Min.	Typ.	Max.
A			1.60			0.063
A1	0.05		0.15	0.002		0.006
A2	1.35	1.40	1.45	0.053	0.055	0.057
b	0.22	0.32	0.38	0.009	0.013	0.015
C	0.09		0.20	0.004		0.008
D		16.00			0.630	
D1		14.00			0.551	
E		16.00			0.630	
E1		14.00			0.551	
e		0.65			0.026	
K	0°	3.5°	0.75°	0°	3.5°	0.75°
L	0.45	0.60	0.75	0.018	0.024	0.030
L1		1.00			0.039	

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