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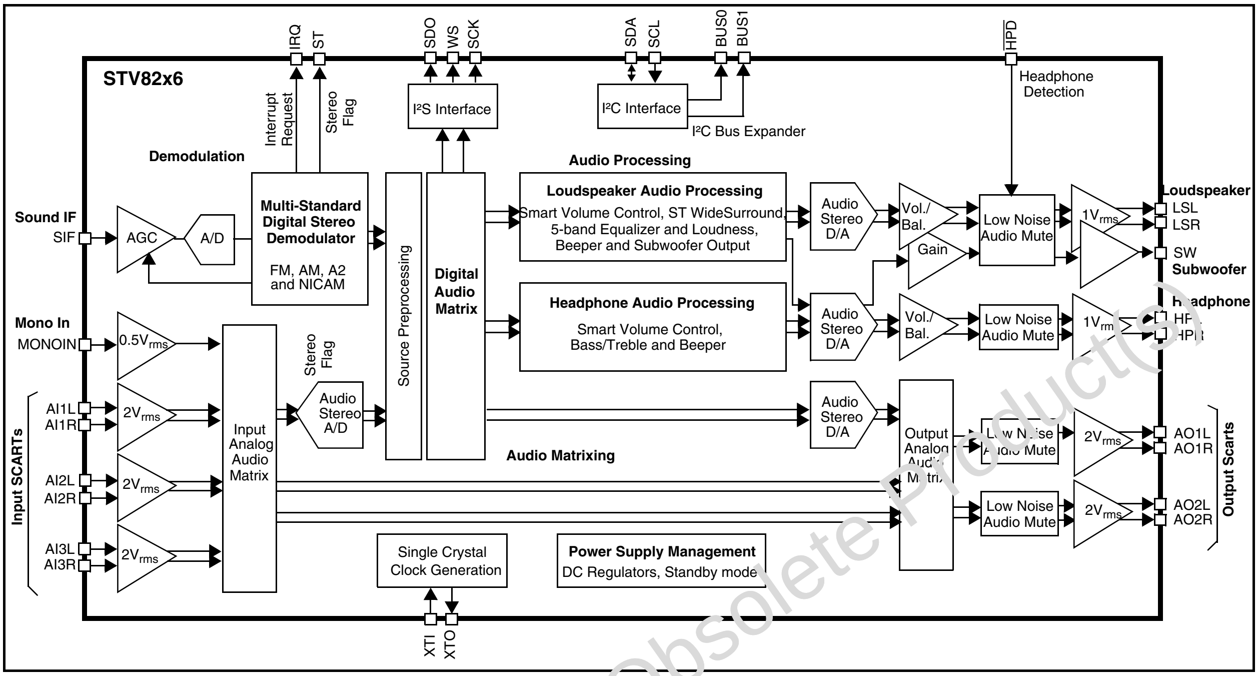
STV82x6

Multistandard TV Audio Processor and Digital Sound Demodulator

DATASHEET



This device incorporates the SRS (Sound Retrieval System) under licence from SRS Labs, Inc.

Key Features

- NICAM, AM, FM Mono and FM 2 Carrier Stereo Demodulators for all sound carriers between 4.5 and 7 MHz
- Mono input provided for optimum AM Demodulation performances
- Demodulation controlled by Automatic Standard Recognition System
- Sound IF AGC with wide range
- Overmodulation and Carrier Offset recovery
- Smart Volume Control
- 5-band Equalizer & Bass/Treble Control
- Automatic Loudness Control
- Loudspeaker and Headphone outputs with Volume/Balance Controls and Beeper
- Subwoofer output with Volume Control and Programmable Bandwidth
- Spatial Sound Effects (ST WideSurround and Pseudo-Stereo)
- SRS® 3D Surround
- 3-to-2 Analog Stereo Audio I/Os (SCART compatible) with Audio Matrix
- Low-noise Audio Mutes and Switches
- I²S Output to interface with Dolby® Pro Logic® Decoder
- I²C Bus-controlled
- Single and standard 27 MHz Crystal Oscillator
- Power supplies: 3.3 V Digital, 5 V or 8 V Analog
- Embedded 3.3 V Regulators
- Packages: SDIP56 or TQFP80

Table of Contents

1 General Description

1.1 Overview

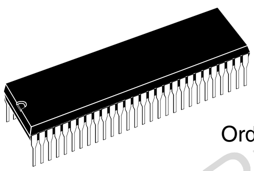
The STV82x6 is composed of three main parts:

1. **TV Sound Demodulator**: provides all the necessary circuitry for the demodulation of audio transmissions of European and Asian terrestrial TV broadcasts. The various transmission standards are automatically detected and demodulated without user intervention.
2. **Audio Processor**: based on DSP technology, independently controls loudspeaker, subwoofer and headphone signals. It offers basic and advanced features, such as a ST WideSurround, Equalizer, Automatic Loudness and Smart Volume Control for television viewer comfort. The STV8226/36 versions can perform additionally the SRS® 3D Surround for stereo and mono signals.
3. **Audio Matrix**: 3 stereo and 1 mono external analog audio inputs to loudspeakers and headphone, with 2 stereo external analog audio outputs (SCART compatible).

Table 1: STV82x6 Version List

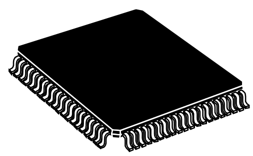
Feature	STV8206	STV8216	STV8226	STV8236
AM-FM Mono	X	X	X	X
Zweiton	X	X	X	X
NICAM		X		X
ST WideSurround	X	X	X	X
SRS® 3D Surround			X	X

Figure 1: Package Ordering Information



SDIP56 Package
Order Code: STV82x6D

TQFP80 Package
Order Code: STV82x6 (Tray)
STV82x6T (Tape & Reel)



1.2 Typical Applications

Figure 2: Typical Application (Low-cost Stereo TV)

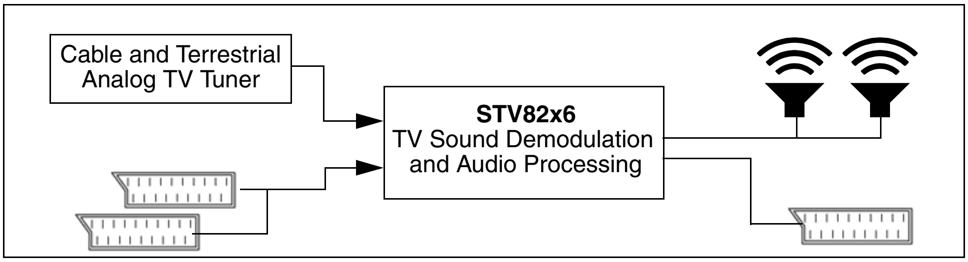


Figure 3: Typical Application with Subwoofer and Headphone

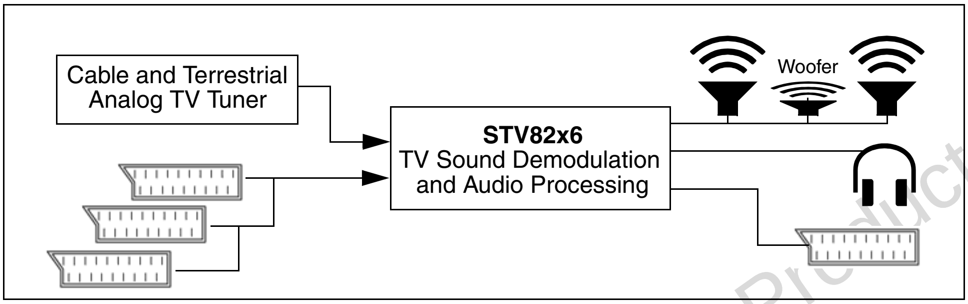


Figure 4: Typical Application Electrical Diagram for STV82x6 in SDIP56 package

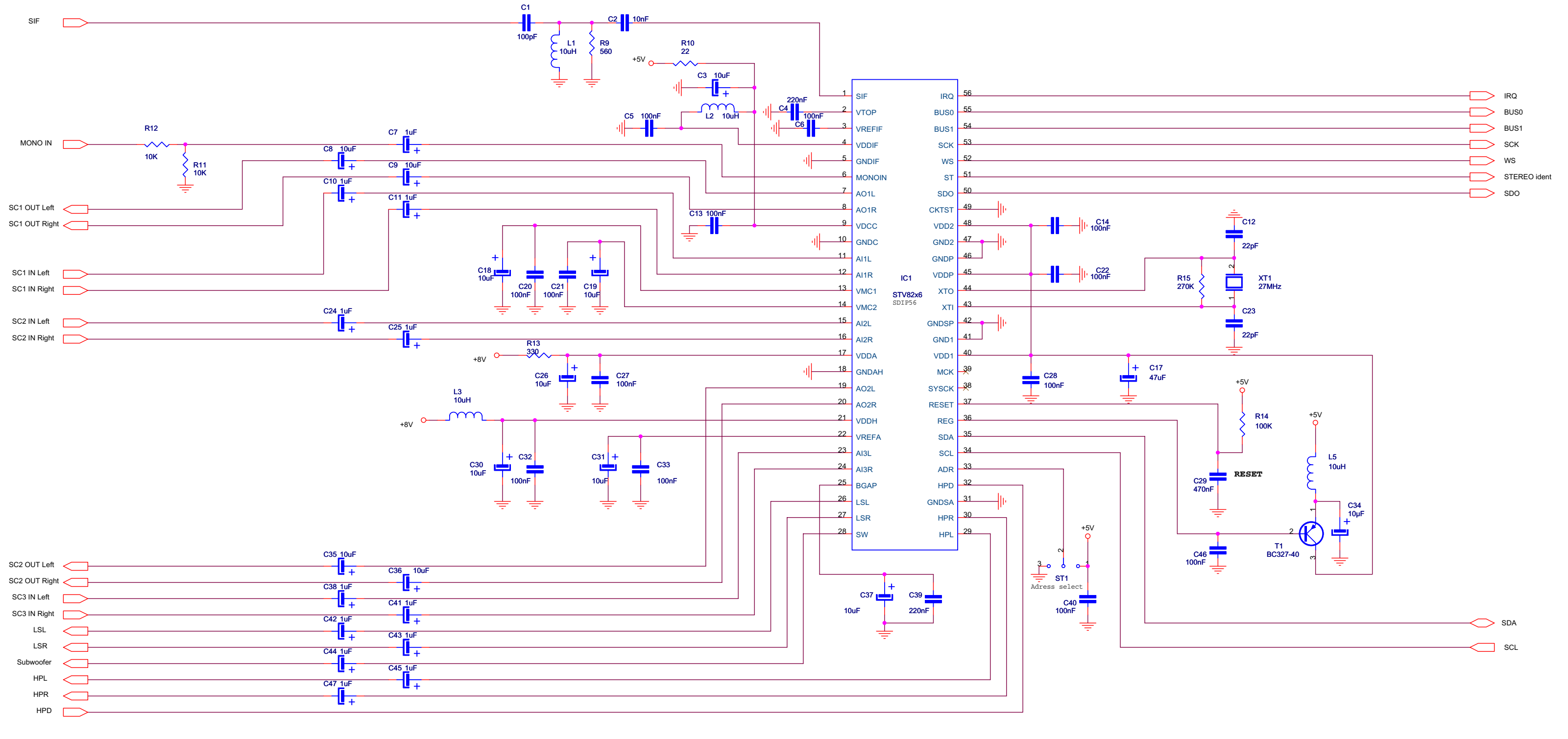


Figure 5: Typical Application Electrical Diagram for STV82x6 in TQFP80 package

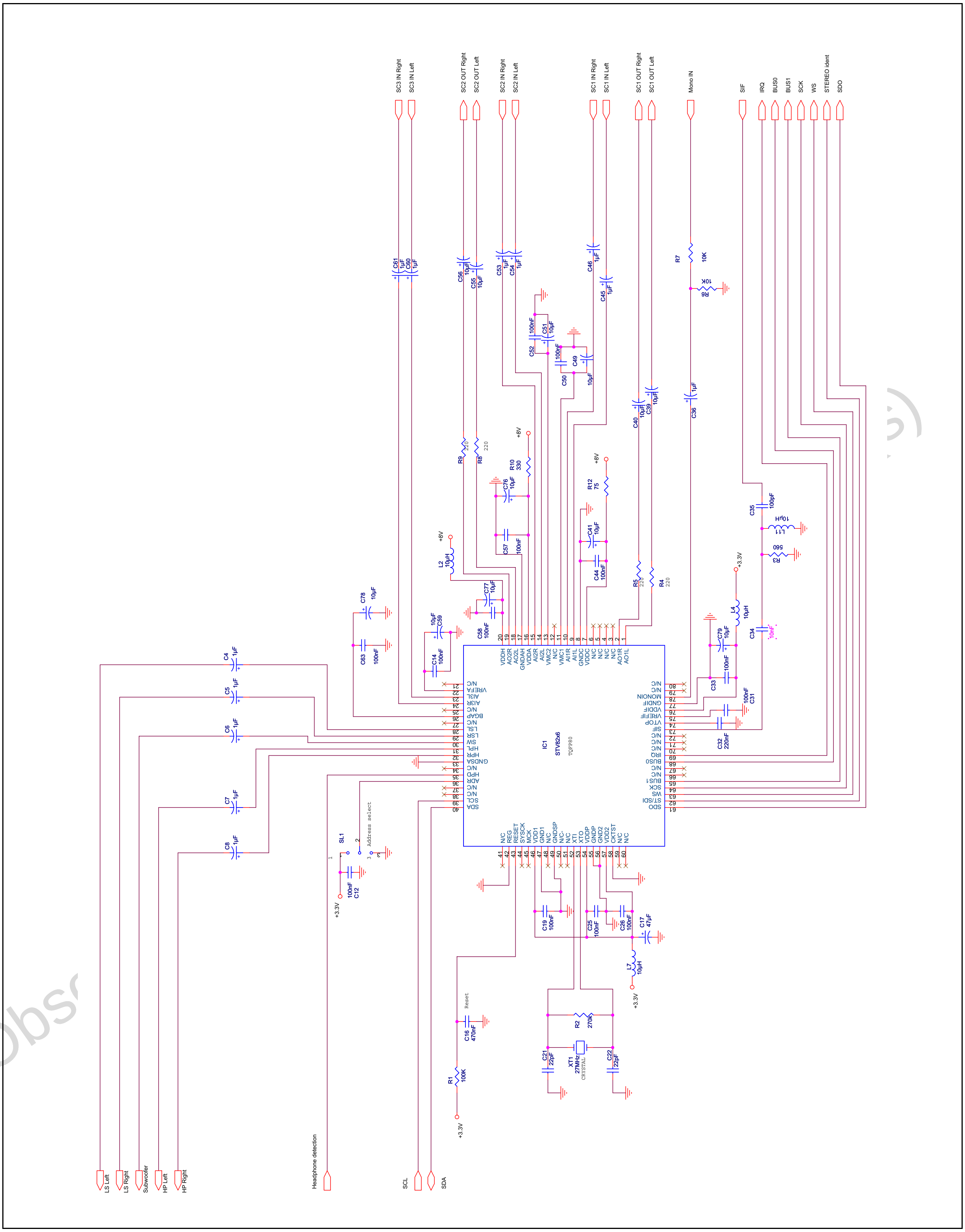
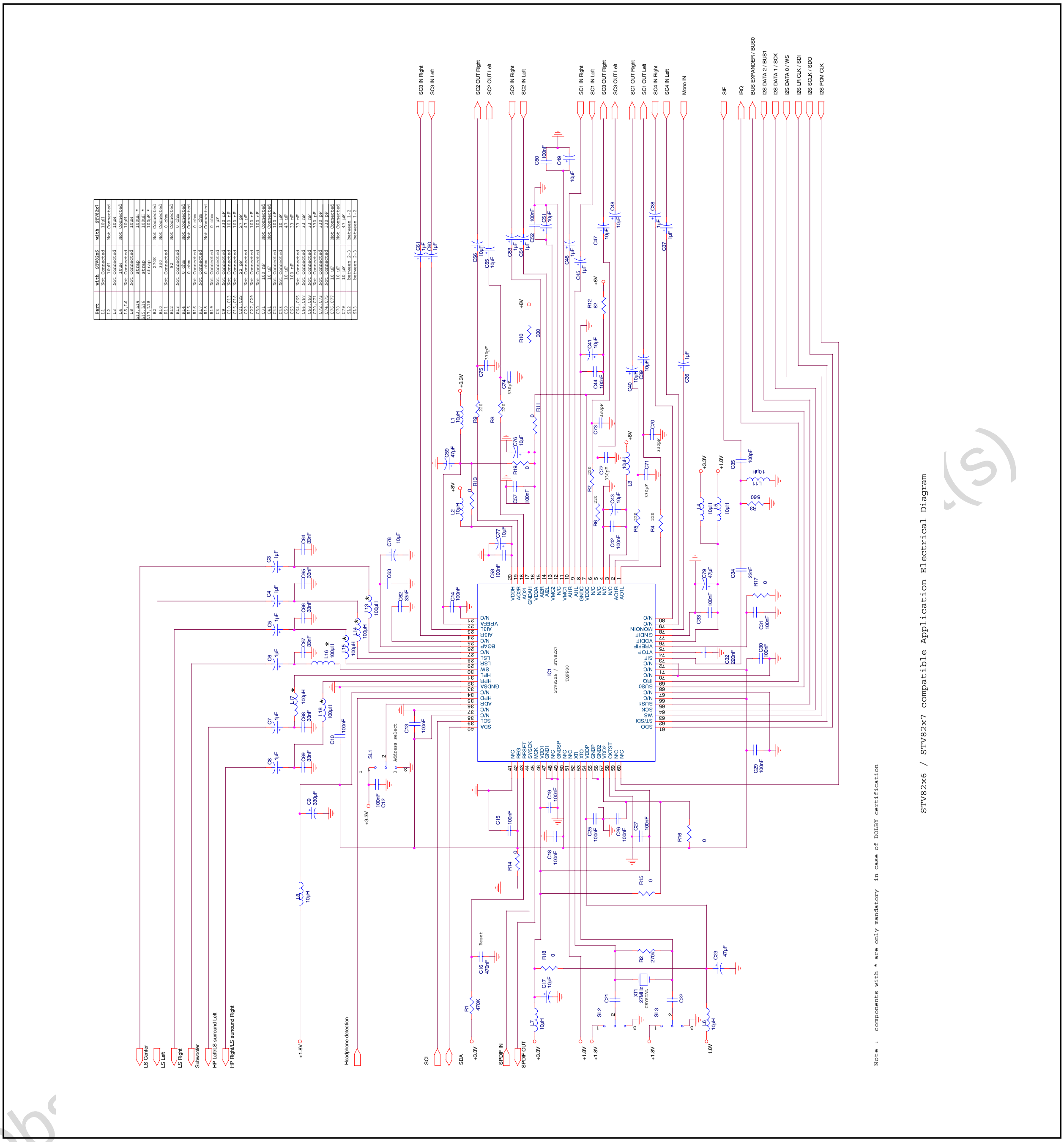


Figure 6: Typical Compatible Application Electrical Diagram for STV82x6 and STV82x7 in TQFP80 package



Part	with STV82x6	with STV82x7
L1	Not Connected	10µH
L2	10µH	Not Connected
L3	Not Connected	10µH
L4	10µH	Not Connected
L5,L6	Not Connected	10µH
L8	Not Connected	10µH
L13,L14	strap	100µH *
L15,L16	strap	100µH *
L17,L18	strap	100µH *
R2	270K	Not Connected
R10	330	Not Connected
R11	Not Connected	0 ohm
R12	82	Not Connected
R13	Not Connected	0 ohm
R14	0 ohm	Not Connected
R15	0 ohm	Not Connected
R16	Not Connected	0 ohm
R17	Not Connected	0 ohm
R18	0 ohm	Not Connected
R19	Not Connected	0 ohm
C3	Not Connected	1 µF
C9	Not Connected	330 µF
C10,C13	Not Connected	100 nF
C15,C18	Not Connected	100 nF
C21,C22	22 pF	27 pF
C23	Not Connected	47 µF
C27,C29	Not Connected	100 nF
C30	Not Connected	100 nF
C31	100 nF	Not Connected
C41	10 µF	Not Connected
C42	Not Connected	100 nF
C43	Not Connected	10 µF
C59	10 µF	47 µF
C63	100 nF	33 nF
C64,C65	Not Connected	33 nF
C66,C67	Not Connected	33 nF
C68,C69	Not Connected	33 nF
C70,C71	Not Connected	330 pF
C72,C73	Not Connected	330 pF
C74,C75	Not Connected	330 pF
C76,C77	10 µF	Not Connected
C78	10 µF	Not Connected
C79	10 µF	47 µF
SL2	between 2-3	between 1-2
SL3	between 2-3	between 1-2

1.3 I/O Pin Description

Legend / Abbreviations for Table 2:

Type:

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

Table 2: Pin Description

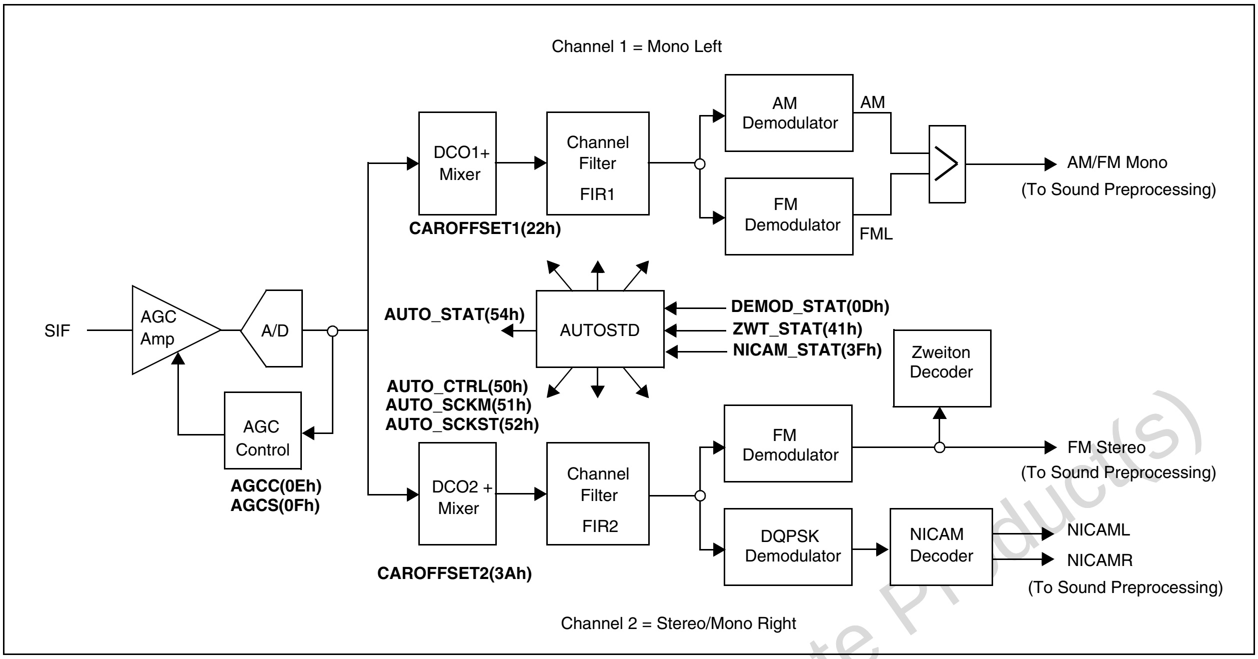
SDIP 56	TQFP 80	Name	Type	Function
1	73	SIF	A	Sound IF Input
2	74	VTOP	A	ADC V_{TOP} Decoupling Pin
3	75	VREFIF	A	AGC Voltage Reference Decoupling Pin
4	76	VDDIF	AP	3.3 V Power Supply for IF AGC & ADC
5	77	GNDIF	AP	0 V Power Supply for IF AGC & ADC
6	78	MONOIN	A	Mono Input
	79/80	N/C		Not Used
7	1	AO1L	A	Left SCART1 Audio Output
8	2	AO1R	A	Right SCART1 Audio Output
-	3/4/5/6	N/C		Not used
9	7	VDDC	AP	3.3 V Power Supply for Audio DAC/ADC
10	8	GNDC	AP	0 V Power Supply for DAC/ADC
11	9	AI1L	A	Left SCART1 Audio Input
12	10	AI1R	A	Right SCART1 Audio Input
13	11	VMC1	A	Switched V_{REF} Decoupling Pin for Audio Converters (VMCP)
-	12	N/C		Not used
14	13	VMC2	A	V_{REF} Decoupling Pin for Audio Converters (VMC)
15	14	AI2L	A	Left SCART2 Audio Input
16	15	AI2R	A	Right SCART2 Audio Input
17	16	VDDA	AP	3.3 V Power Supply for Audio Buffers, Matrix & Bias
18	17	GNDAH	AP	0 V Power Supply for Audio Buffers & SCART
19	18	AO2L	A	Left SCART2 Audio Output
20	19	AO2R	A	Right SCART2 Audio Output
21	20	VDDH	AP	8 V / 5 V Power Supply for SCART & Audio Buffers
-	21	N/C		Not Used
22	22	VREFA	A	Voltage Reference for Audio Buffers
23	23	AI3L	A	Left SCART3 Audio Input
24	24	AI3R	A	Right SCART3 Audio Input
-	25	N/C		Not Used
25	26	BGAP	A	Bandgap Voltage Source Decoupling

Table 2: Pin Description (Continued)

SDIP 56	TQFP 80	Name	Type	Function
-	27	N/C		Not Used
26	28	LSL	A	Left Loudspeaker Output
27	29	LSR	A	Right Loudspeaker Output
28	30	SW	A	Subwoofer Output
29	31	HPL	A	Left Headphone Output
30	32	HPR	A	Right Headphone Output
31	33	GNDSA	AP	Substrate Analog/Digital Shield
-	34	N/C		Not Used
32	35	$\overline{\text{HPD}}$	B	Headphone Detection Input (Active Low)
33	36	ADR	I	Hardware I^{2}C Chip Address Control
-	37/38	N/C		Not Used
34	39	SCL	OD	I^{2}C Serial Clock
35	40	SDA	OD	I^{2}C Serial Data
-	41	N/C		Not Used
36	42	REG	A	5 V Power Regulator Control
37	43	$\overline{\text{RESET}}$	I	Hardware Reset (Active Low)
38	44	SYSCK	B	System Clock Output
39	45	MCK	B	I^{2}S Master Clock Output
40	46	VDD1	DP	3.3V Power Supply for Digital Core & IO Cells
41	47	GND1	DP	0V Power Supply for Digital Core & IO Cells
-	48	N/C		Not Used
42	49	GNDSP	AP	Substrate Analog/Digital Shield for Clock-PLL
	50/51	N/C		Not Used
43	52	XTI	I	Crystal Oscillator Input
44	53	XTO	O	Crystal Oscillator Output
45	54	VDDP	AP	3.3 V Power Supply for Analog PLL Clock
46	55	GNDP	AP	0 V Power Supply for Analog PLL Clock
47	56	GND2	DP	0 V Power Supply for Digital Core, DSPs & IO Cells
48	57	VDD2	DP	3.3 V Power Supply for Digital Core, DSPs & IO Cells
49	58	CKTST	I	Must be Connected to 0 V
-	59/60	N/C		Not Used
50	61	SDO	B	I^{2}S Bus Data Output
51	62	ST/SDI	B	Stereo Detection Output / I^{2}S Bus Data Input
52	63	WS	B	I^{2}S Bus Word Select Output
53	64	SCK	B	I^{2}S Bus Clock Output
54	65	BUS1	B	I^{2}C Bus Expander Output 1
-	66/67	N/C		Not Used
55	68	BUS0	B	I^{2}C Bus Expander Output 2
56	69	IRQ	B	I^{2}C Status Read Request
-	70	N/C		Not Used
-	71	N/C		Not Used
-	72	N/C		Not Used

2 Demodulator Block

Figure 7: Demodulator Block Diagram



Note: Zweiton is the Dual (Two Tone) FM stereo or A2 system.

2.1 Digital Demodulator

2.1.1 Sound IF Signal

The Analog Sound Carrier IF is connected to STV82x6 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 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 transmitted audio signal. A preset I²C parameter is required to define the gain of the AGC used in Manual mode (Registers AGCC and AGCS).

2.1.2 Demodulation

The demodulation system operates by default in Automatic mode. In this mode, the STV82x6 is able to **identify and demodulate any TV sound standard including NICAM and A2 systems** (see Table 2) without any external control via the I²C 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** (AUTOSTD) automatically programs the appropriate bits in the I²C registers which are forced to Read-only mode for users (see Section 9.1). 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 AUTO_SCKM and AUTO_SCKST, respectively. The identified standard is displayed in register AUTO_STAT 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 AUTO_CTRL while checking the detected standard status by reading registers AUTO_STAT, NICAM_STAT and ZWT_STAT. Moreover, the detection of Stereo mode during demodulation is also flagged in register AUTO_STAT and on output pin ST.

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 programmed in register AUTO_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 I²C controls are provided without interfering with the Automatic Standard Recognition System (AUTOSTD).

DK-NICAM Overmodulation Recovery: Four different FM deviation ranges can be selected (via register AUTO_CTRL) for the DK standard while the AUTOSTD 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_STATL and PEAK_DET_STATR. This value is used to implement Automatic Overmodulation Detection via an external I²C 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 AUTO_SCKM and AUTO_SCKST) when programming larger FM deviations reserved only for DK-NICAM standards.

Table 3: Standards covered by the Automatic Standard Recognition System (AUTOSTD)

System	Sound Type	Type Name	Carrier 1 (MHz)	Carrier 2 (MHz)	FM/AM Deviation			De-emphasis	Roll-off (%)	Pilot Frequency (kHz)
					Min.	Typ.	Max.			
M/N	FM Mono		4.5		15	27	50	75 µs		
	FM 2 Carriers	A2+		4.724						55.069
B/G	FM Mono		5.5		27	50	80	50 µs		
	FM/NICAM			5.850				J17	40	
	FM 2 Carriers	A2		5.742				50 µs		54.6875
I	FM Mono		6.0		27	50	80	50 µs	100	
	FM/NICAM			6.552				J17	100	
L	AM Mono		6.5		0.5		1.0			
	AM/NICAM			5.850				J17	40	
D/K	FM Mono				27	50	80	50 µs		
	FM/NICAM			5.850				J17	40	
D/K1	FM 2 Carriers	A2*		6.258				50 µs		54.6875
D/K2	FM 2 Carriers			6.742						
D/K3	FM 2 Carriers			5.742						

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 AUTOSTD system remains active. The frequency offset estimation is written in registers FM_DCL and FM_DCR (Mono Left / Channel 1 And Mono Right / Channel 2, respectively) and can be used to implement the Automatic Frequency Control (AFC) via an external I²C control.

If required, the AUTOSTD system can be disabled (Manual mode) and the user can control all registers including those only controlled by the AUTOSTD function when active. Manual mode is selected in registers RESET or AUTO_SCKM.

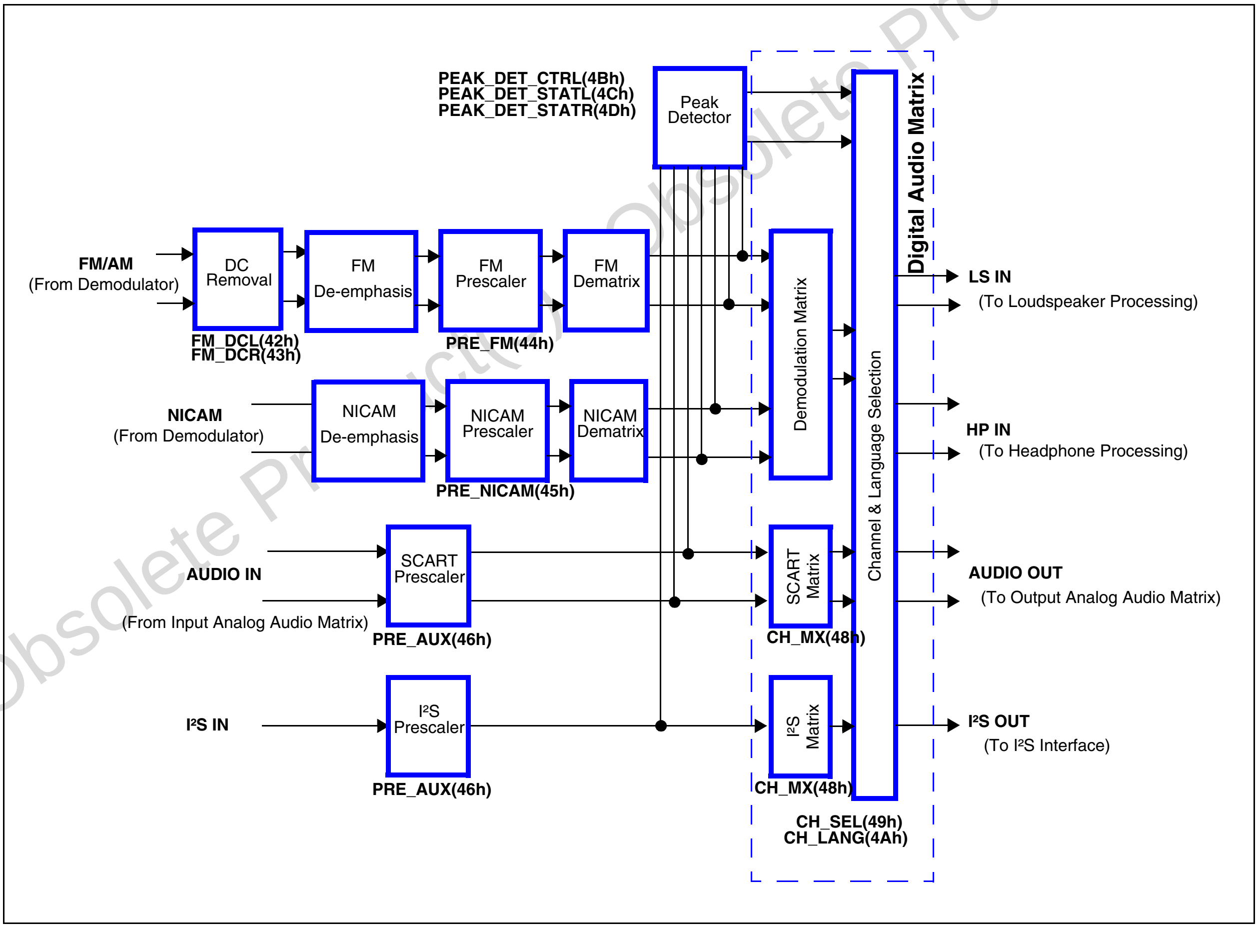
2.1.3 Sound Preprocessing and Selection

The demodulated sound signal can be redirected to 4 different output audio channels:

1. Loudspeaker & Subwoofer,
2. Headphone,
3. SCART,
4. I²S Interface.

Each output channel can independently select the demodulator source, analog SCART or I²S inputs using register CH_SEL.

Figure 8: Sound Preprocessing and Selection Block Diagram



The level of the demodulated sound may require adjusting in order to compensate for the difference in levels between the multiple source (NICAM, FM or AM) and standard source (FM deviation wide range from 15 to 500 kHz) signals. The correct range for all level variations (+24 to -6 dB) is selected in registers PRE_FM and PRE_NICAM. The internal sound level of the various sources (FM/AM, NICAM and SCART) is read in registers PEAK_DET_CTRL, PEAK_DET_STATL and PEAK_DET_STATR before audio processing and can be used to implement Automatic Pre-scaling via an external I^2C interface.

In Automatic mode, the STV82x6 selects and performs all appropriate de-emphasis, dematrixing, sound selection and mute functions according to the standard and transmission mode detected.

Mono system: Mono audio signals received by an FM or AM carrier are demodulated. Left and right audio outputs are identical. Automatic mute is applied when the mono standard cannot be identified.

A2 systems (or Zweiton): Transmission of mono, stereo or bilingual audio signals using 2 separate FM carriers + identification pilot. The pilot, transmitted by the second carrier, can be modulated by two different tones in order to define Stereo or Dual-Mono mode. If not modulated, only the mono signal is broadcast on the first carrier. Zweiton mode is read in register ZWT_STAT and described in Table 4. In the event of poor signal detection, the audio output is switched back to FM Mono mode (backup). In Dual Mono mode, the language (A on Channel 1, B on Channel 2) can be selected separately for each audio output channel (Loudspeaker, Headphone, SCART or I^2S) in register CH_LANG.

Table 4: A2 System Transmission Modes

System Mode	ZWT-STAT [2:0]	FM Dematrix	FM De-emphasis	CH_LANG [1:0]	Sound Selection	Sound Backup
German Zweiton Mono	100	L,R	50 µs	XX	FM Mono	X
German Zweiton Stereo	110	(L+R)/2,R	50 µs	XX	FM Stereo	FM Mono
German Zweiton Dual Mono (CH1=A, CH2=B)	101	L,R	50 µs	01	FM Mono A	X
				10	FM Mono B	Mute
Korean Zweiton Mono	100	L,R	75 µs	XX	FM Mono	X
Korean Zweiton Stereo	110	(L+R)/2,(L-R)/2	75 µs	XX	FM Stereo	FM Mono
Korean Zweiton Dual Mono (CH1 = A, CH2 = B)	101	L,R	75 µs	01	FM Mono A	X
				10	FM Mono B	Mute
Zweiton undefined	0XX or 111	L,R	50 µs	XX	FM Mono	X

Note: A2 and A2 standards are German Zweiton, while A2+ is Korean Zweiton.*

NICAM systems: Transmission of mono, stereo, bilingual or trilingual audio signals using a modulated-QPSK carrier and an FM/AM sound carrier backup. The digital QPSK modulation broadcasts either channel stereo, dual mono, mono + data or data only. The selected NICAM mode is read in register NICAM_STAT and described in Table 5. In the event of high bit-error rates, the audio output is automatically switched back to the reserve sound transmission (FM/AM Mono) or muted if there is no backup. In Dual Mono or Stereo mode with no backup, the language can be selected separately for each audio output channel (Loudspeaker, Headphone, SCART or I^2S) in register CH_LANG.

Table 5: NICAM System Transmission Modes

System Mode	NICAM_STAT[4:1]	NICAM De-emphasis	CH_LANG[1:0]	Sound Selection	Sound Backup
NICAM Stereo	1000	J17	XX	NICAM Stereo	FM/AM Mono
NICAM Dual Mono (CH1 = A, CH2 = B)	1010	J17	01	NICAM Mono A	FM/AM Mono
			10	NICAM Mono B	Mute
NICAM Mono+Data (D1 = A, D2 = Data)	1001	J17	XX	NICAM Mono A	FM/AM Mono
NICAM Data	1011	J17	XX	FM/AM Mono	X
NICAM Stereo (no backup)	0000	J17	01	FM/AM Mono A	X
			00	NICAM Stereo	Mute
NICAM Dual Mono (no backup) (D1 = B, D2 = C)	0010	J17	01	FM/AM Mono A	X
			10	NICAM Mono B	Mute
			11	NICAM Mono C	
NICAM Mono+Data (no backup) (D1 = B, D2 = Data)	0001	J17	01	FM/AM Mono A	X
			10	NICAM Mono B	Mute
NICAM undefined (no backup)	X1XX	J17	XX	FM/AM Mono	X

Note: D1 and D2 define the two channels encoded in the NICAM packet.

2.2 System Clock

The System Clock integrates a low-jitter PLL clock and can be fully reprogrammed via registers PLL_DIV, PLL_MD, PLL_PEH and PLL_PEL. The default values are designed for **a standard 27-MHz quartz crystal frequency**, which is the recommended frequency for minimizing potential RF interference in the application. This sinusoidal clock frequency, and any harmonic products, remains outside the TV picture and sound IF (PIF/SIF) and Band-I RF passbands and has been selected in order to reduce the risk of potential interference to the TV IF and RF system.

However, if required, the PLL clock can be re-programmed for an other quartz crystal frequency within a range between 23 and 30 MHz.

Note: A change in the crystal frequency is compatible with other default I²C programming values, including those of the built-in Automatic Standard Recognition System.

3 Audio Processor Block

3.1 Main Features

The STV82x6 Audio Processor is based on a dedicated audio Digital Signal Processor (DSP) that performs basic and advanced audio post-processing for 4 different output audio channels.

3.1.1 Loudspeaker and Subwoofer Features

- Smart Volume Control (See Note 1)
- Spatial effects:
  - — Pseudo Stereo (for Mono source)
  - — ST WideSurround (“Movie” and “Music” modes for Stereo source)
- 5-band Equalizer
- Volume and Balance controls (See Note 4)
- Automatic Loudness control
- Subwoofer (See Note 4)
- Beeper (See Note 3)

Additionally on STV8226/36 only:

- SRS™ 3D Mono signal processing
- SRS™ 3D Stereo signal processing

3.1.2 Headphone (See Note 2)

- Smart Volume Control (See Note 1)
- Bass and Treble controls
- Volume and Balance controls
- Beeper (See Note 3)

Note: 1 The Smart Volume Control can be used in either the loudspeaker or headphone path, but not both at the same time.

2 The headphone is forced into Mono mode when the subwoofer is active.

3 The beeper is common for both the loudspeaker and the headphone.

4 The Auto-mute function is activated when a headphone plug is detected.

5 All audio postprocessing can be disabled.

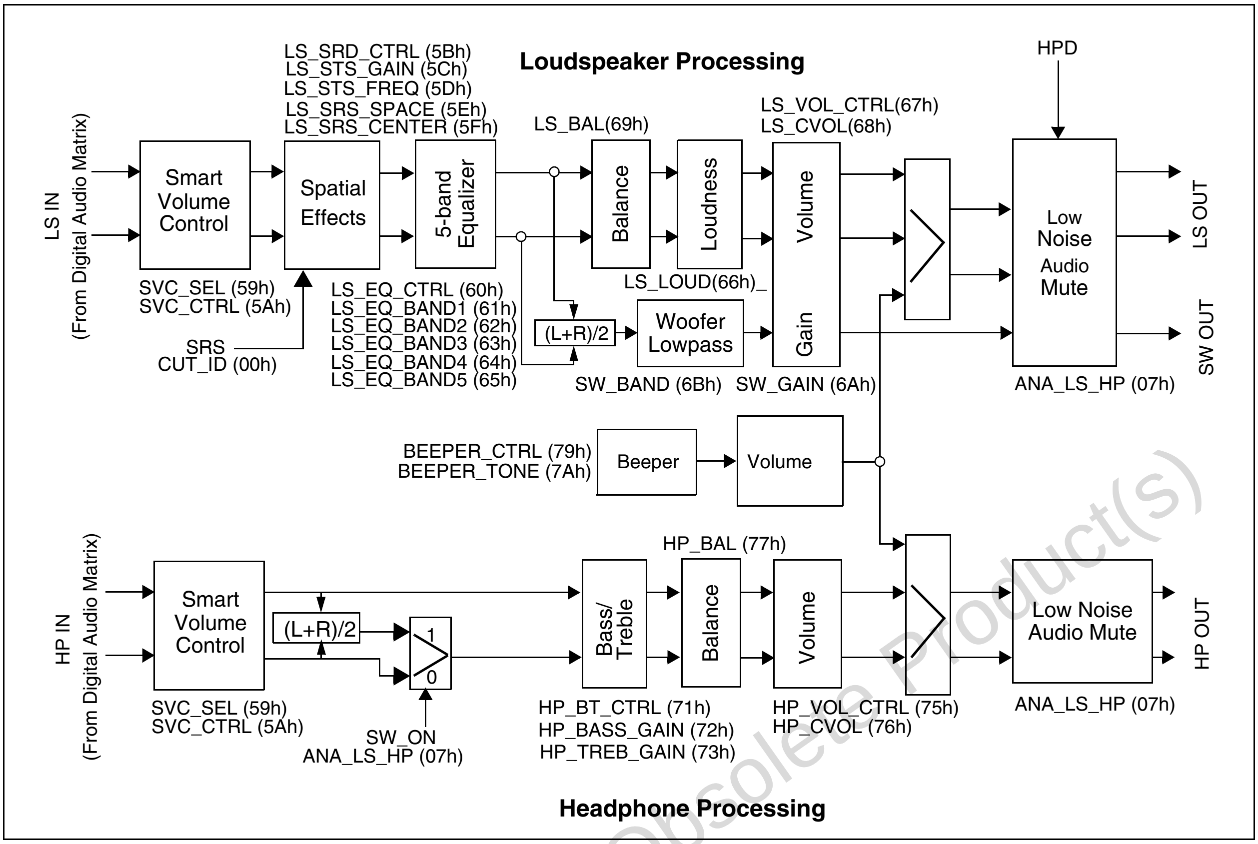
3.1.3 SCART 1 and 2 Outputs

- No audio post-processing

3.1.4 I²S Output

- No audio post-processing

Figure 9: Audio Processor Block Diagram



Note: The audio signals available on the I²S and SCART outputs are not affected by any digital or analog matrix processing.

3.2 Smart Volume Control (SVC)

The Smart Volume Control (SVC) feature is designed to process sound level variations caused by changes in signal sources (e.g. when switching channels) or in volume (e.g. when advertisements are broadcast). The SVC is controlled by the SVC_ON bit in the SVC_CTRL register.

When the SVC_ON bit is set, the Smart Volume Control prevents annoying volume changes by automatically adjusting the selected sound source (demodulator or SCART) to a programmable reference level before audio processing. The regulation ranges from +6 dB to -30 dB with a fast attenuation and a programmable slow amplification. The fast attenuation reduces audio peak (and potential clipping) and slow amplification is a compromise between regulation recovery and limited audio amplification during audio silence. The programmable output reference level must be defined to prevent internal clipping depending on the selected audio processing boosting functions such as Surround (up to +9 dB), Equalizer or Bass/Treble (up to +12 dB) and Loudness (up to +6 dB). When the SVC is enabled, recommended reference values are -18 dB for the Loudspeaker path and -9 dB for the Headphone path.

When the SVC is disabled, it acts as a wide-range prescaler (between -30 dB and +15.5 dB) before audio-processing to prevent internal clipping depending on the selected functions (see above). If required, it complements the dedicated prescaler for FM, NICAM or SCART sources. The internal level can be measured using the peak detector.

The SVC can be used either in the Loudspeaker or Headphone path (but not both simultaneously). When used in the Headphone path, the SVC prevents the sound level from becoming suddenly too strong, causing ear damage. The SVC is configured in registers SVC_SEL and SVC_CTRL.

3.3 ST WideSurround

STV82x6 offers three preset ST WideSurround effects on the Loudspeaker path:

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

"ST WideSurround" 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.

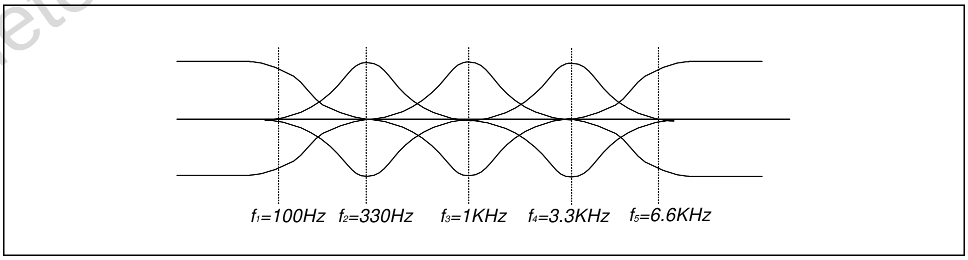
The Surround/Pseudo-stereo mode is automatically selected by the Automatic Standard Recognition System (AUTOSTD) depending on the detected stereo or mono source. By default, "Movie" is selected for Surround mode. This value may be changed to "Music" by the STS_MODE bit in the LS_SRD_CTRL register.

Additional user controls are provided to better adapt the spatial effect to the source. The ST WideSurround Gain (LS_STS_GAIN) and ST WideSurround Frequency (LS_STS_FREQ) registers can be used to enhance music predominance in Music mode and theater effect + voice predominancy in Movie mode.

3.4 5-Band Audio Equalizer

The Loudspeaker audio spectrum is split into 5 frequency bands and the gain of each of them can be adjusted within a range from -12 dB to +12 dB in steps of 1 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 EQ_ON bit in the LS_EQ_CTRL register. The Bass, Medium and Treble values are programmed in registers LS_EQ_BAND[1:5].

Figure 10: Equalizer



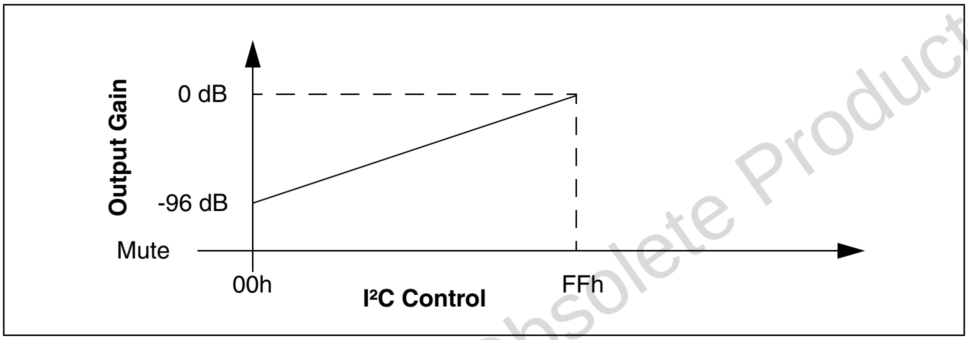
3.5 Bass/Treble Control

The gain of bass and treble frequency bands for the headphone can be also tuned within a range from -12 dB to +12 dB in steps of 1 dB. It may be used to pre-define frequency band enhancement features dedicated to various kinds of music, to implement programmable Loudness or Super-bass functions. The Headphone Bass/Treble feature is enabled by setting the BT_ON bit in the HP_BT_CTRL register. The Bass and Treble gain values are adjusted in registers HP_BASS_GAIN and HP_TREBLE_GAIN, respectively.

3.6 Volume/Balance Control

The STV82x6 provides a Volume/Balance Control for each of the Loudspeaker, Subwoofer and Headphone audio outputs. Its wide range (from 0 to -96 dB in a linear scale) largely covers typical home applications (approx. 60 dB) while maintaining a good S/N ratio. Its fine resolution (0.375 dB) provides simple volume programming and a relative OSD scale representation. The Loudspeaker, Subwoofer and Headphone volume values should be programmed progressively in steps of less than 1 dB in order to prevent audible envelope variations and a minimum duration of 16 ms is required between two successive programming commands to guarantee that there are no audible plops during volume changes. In this case, a full 8-bit volume scan with minimum steps of 0.375 dB will last approximately 4 s (minimum).

Figure 11: Volume Control



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

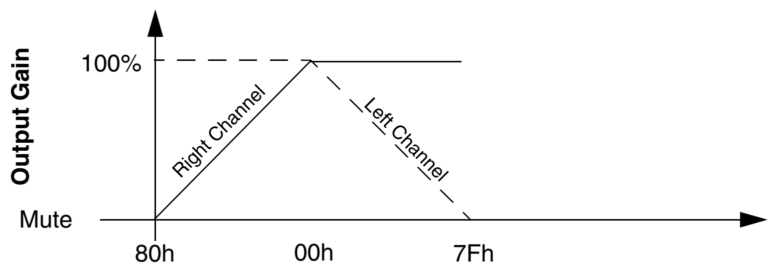
- In **Differential mode** (default value), the volume control is a common volume value for both the Left and Right Loudspeaker and Headphone channels.
- In **Independent mode**, the volume for the Left and Right channels for Loudspeakers or Headphone is controlled independently.

As the Loudspeaker bass frequencies are output by the Subwoofer, its reference volume is controlled by default with the value of the LS_CVOL common volume register. The SW_GAIN register value is used to adjust the level of the Subwoofer output in regards to this reference. In Independent mode, the SW_GAIN register is used as a separated volume control and does not take into account the Loudspeaker audio level.

3.6.1 Differential Mode

The common value for the Right/Left volume controls for the Loudspeaker, Subwoofer and Headphone outputs are programmed in registers LS_CVOL, SW_GAIN and HP_CVOL, respectively. A differential balance can be applied using registers LS_BAL and HP_BAL to adjust the Left/Right level ratio as shown in Figure 12.

Figure 12: Differential Balance



3.6.2 Independent Mode

This is enabled by setting the BAL_MODE bits in both the LS_VOL_CTRL and HP_VOL_CTRL registers to Independent mode. In this case, the register values are used to control the volume/balance functions as described in Table 6.

Table 6: Volume/Balance Control Registers

Mode	LS_CVOL/LS_VOL_L HP_CVOL/HP_VOL_L Register 68h/76h	LS_BAL/LS_VOL_R HP_BAL/HP_VOL_R Register 69h/77h
LS_VOL_CTRL (Loudspeaker Volume Control)		
BAL_MODE = 0 (Independent Mode)	LS_VOL_L Left Volume value	LS_VOL_R Right Volume value
BAL_MODE = 1 (Differential Mode)	LS_CVOL Common Right/Left Volume value	LS_BAL Differential Balance value
HP_VOL_CTRL (Headphone Volume Control)		
BAL_MODE = 0 (Independent Mode)	HP_VOL_L Left Volume value	HP_VOL_R Right Volume value
BAL_MODE = 1 (Differential Mode)	HP_CVOL Common Right/Left Volume value	HP_BAL Differential Balance value

3.6.3 Mute Control

An Independent Mute Control can be used to smooth audio envelope variations in order to prevent any audible plops can be applied to all audio outputs. This feature is controlled by register ANA_LS_HP.

A Headphone Detection Mode that will automatically mute the Loudspeaker and Subwoofer outputs when a headphone is detected can be enabled by the HDP_ON bit in the ANA_LS_HP register. In this case, only the Headphone output will remain active. See also Section 3.8: Subwoofer Control and Section 5.4: Headphone Detection.

When a demodulated source is selected on the audio output, the mute is also controlled by Automatic Standard Recognition system (AUTOSTD). In case of no mono detected or bad detection of language without backup, the corresponding audio output is automatically muted. In case of multi-language, the output will be de-muted by selecting an other language with backup.

Table 7: Headphone/Mute Register Configuration

ANA_LS_HP Register						Output Status	
HPD_IN	HPD_ON	SW_ON	MUTE_LS	MUTE_SW	MUTE_HP	Muted	Active
X	0	0	0	X	0	SW	LS, HP Stereo
X	X	1	0	0	1	HP	LS & SW
X	X	X	1	1	1	LS, SW & HP (Channel Change: Mute All)	
X	0	1	0	0	0		LS, SW & HP Mono
0	1	0	0	0	0	SW & HP	LS (Default)
1	1	0	0	0	0	SW & LS	HP Stereo

3.7 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). 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 Loudspeaker Loudness function is enabled by setting the LOUD_ON bit in register LS_LOUD. The Loudness Threshold and Maximum Treble Gain values are also programmed in this register.

Two bass cut-off frequencies are available:

- 40 Hz for Normal mode
- 120 Hz for Bass Amplified mode

The mode is selected by the LOUD_FREQ bit in register LS_LOUD (66h).

3.8 Subwoofer Control

The subwoofer signal is created by adding the bass frequency of the Left/Right Loudspeaker channels. The Subwoofer output is enabled by setting the SW_ON bit in register ANA_LS_HP. This will also force the Headphone output into Mono mode.

The Subwoofer Gain and Frequency Bandwidth values are programmed in registers SW_GAIN and SW_BAND, respectively. The cut-off frequency can be adjusted from between 50 and 400 Hz in steps of 50 Hz.

3.9 Beeper

The beeper is used to replace the audio signal with a tone on the Loudspeaker or Headphone outputs. It can be used for various applications such as beep sounds for remote control, alarm clock or other features.

Obsolete Product(s)

The Beeper operates in one of two modes:

- **Pulse mode** (beep applications) A tone with a programmable short duration (between 128 ms and 1 s) is generated. Afterwards, the beeper is automatically disabled and the output is switched back to the audio signal.
- **Continuous mode** (alarm application) A tone with a programmable long duration is generated. Its start and stop controls must be programmed by I²C.

In both modes, it is recommended to use the mute function to smooth the audio-to-beeper and beeper-to-audio (Continuous mode only) transitions. The second transition is automatically muted in Pulse mode. Beeper parameters are controlled in register BEEPER_CTRL.

The beeper tone level and frequency are programmed in register BEEPER_TONE. The level (or volume) ranges between 0 dB and -93 dB in steps of 3 dB and the tone frequency ranges between 62.2 Hz and 8 kHz in steps of 1 octave.

A beep generator is shared only by the Loudspeaker 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.

Note: The audio output is not affected by the Automatic Mute Control of Automatic Standard Recognition function when the beeper is activated.

Figure 13: Pulse Mode

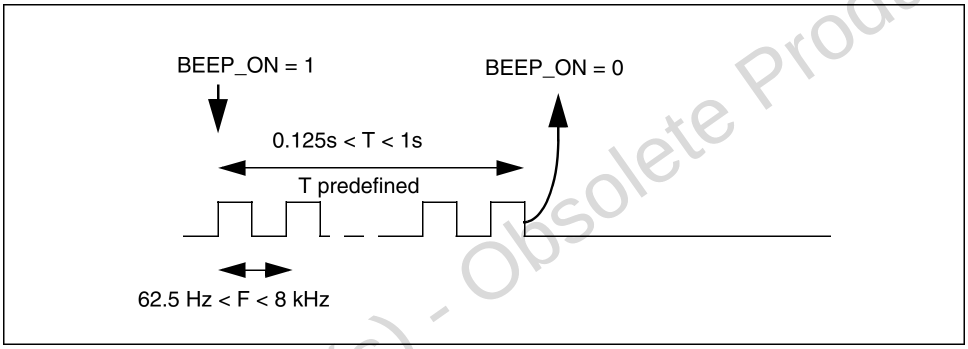
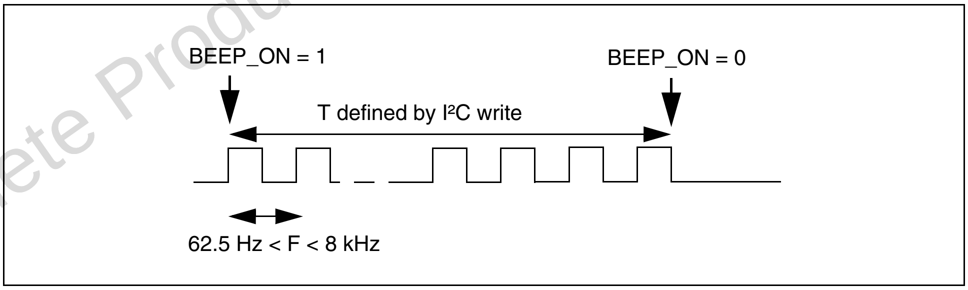


Figure 14: Continuous Mode



3.10 SRS™ 3D Surround (STV8226/36 only)

In addition to ST WideSurround, the STV8226/36 provides SRS™ 3D Stereo and Mono outputs which are spatial effects patented by SRS Labs. The SRS™ system is available on the IC when the SRS_ON bit of register CUT_ID is set (STV8226/36 identification). ST and SRS™ Surround systems cannot be used simultaneously. These signals are output only on the Loudspeaker path.

SRS™ creates a fully immersed three-dimensional soundfield through the use of a standard 2-speaker stereo configuration. For monaural audio, the source is first converted into a synthetic stereo signal before creating the 3D effect. The virtual gain for the Surround and Center components can be adjusted by registers LS_SRS_SPACE and LS_SRS_CENTER (respectively) in Stereo mode only. These values are used to adapt spatial effects to the source.

For ST WideSurround Sound, Stereo or Mono output mode is automatically selected by the Automatic Standard Recognition System (AUTOSTD) according to the detected audio source. By default, ST WideSurround Sound is selected. SRS™ Surround is selected in register LS_SRD_CTRL.

4 Audio Matrices

In addition to the sound carrier source (SIF), the STV82x6 accepts up to three analog stereo audio inputs (2 V_{RMS} SCART compatible) and one analog mono audio input (0.5 V_{RMS}). These different sources can go back out through four analog stereo audio outputs which are Loudspeaker + Subwoofer and Headphone (1 V_{RMS}) and two compatible SCART audio outputs (2 V_{RMS}). An extra digital stereo output (I²S compatible) is available for interfacing with a Dolby Pro Logic Decoder or an external Digital-to-Analog Converter (DAC).

Figure 15: Audio Matrix Block Diagram