

PRELIMINARY DATA SHEET

MSP 44x8G Multistandard Sound Processor

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Multistandard Sound Processor Family

1. Introduction

The MSP 44x8G family of Multistandard Sound Processors covers the sound processing of all analog TV-Standards worldwide, as well as the NICAM digital sound standards. The full TV sound processing, starting with analog sound IF signal-in, down to processed analog AF-out, is performed on a single chip. Fig. 1-1 shows a simplified functional block diagram of the MSP 44x8G.

The high-quality A/D and D/A converters offer the full audio bandwidth of 20 kHz and the backend DSP processing is performed at a 48 kHz sample rate.

The MSP 44x8G has been designed for the usage in hybrid set-top boxes and multimedia applications. Its asynchronous I²S slave interface allows the reception of digital stereo signals with arbitrary sample rates ranging from 5 to 50 kHz. Synchronization is performed by means of an adaptive sample rate converter.

This generation of TV sound processing ICs includes versions for processing the multichannel television sound (MTS) signal conforming to the standard recommended by the Broadcast Television Systems Committee (BTSC). The DBX noise reduction, or alternatively, Micronas Noise Reduction (MNR) is performed alignment free.

Other processed standards are the Japanese FM-FM multiplex standard (EIA-J) and the FM Stereo Radio standard.

The MSP 44x8G versions are pin and software compatible to other MSP families. Standard selection requires only a single I²C transmission.

The MSP 44x8G has built-in automatic functions: The IC is able to detect the actual sound standard automatically (Automatic Standard Detection). Furthermore, pilot levels and identification signals can be evaluated internally with subsequent switching between mono/stereo/bilingual; no I²C interaction is necessary (Automatic Sound Selection).

The ICs are produced in submicron CMOS technology and are available in the following packages: PQFP80, PLQFP64, and PSDIP64.

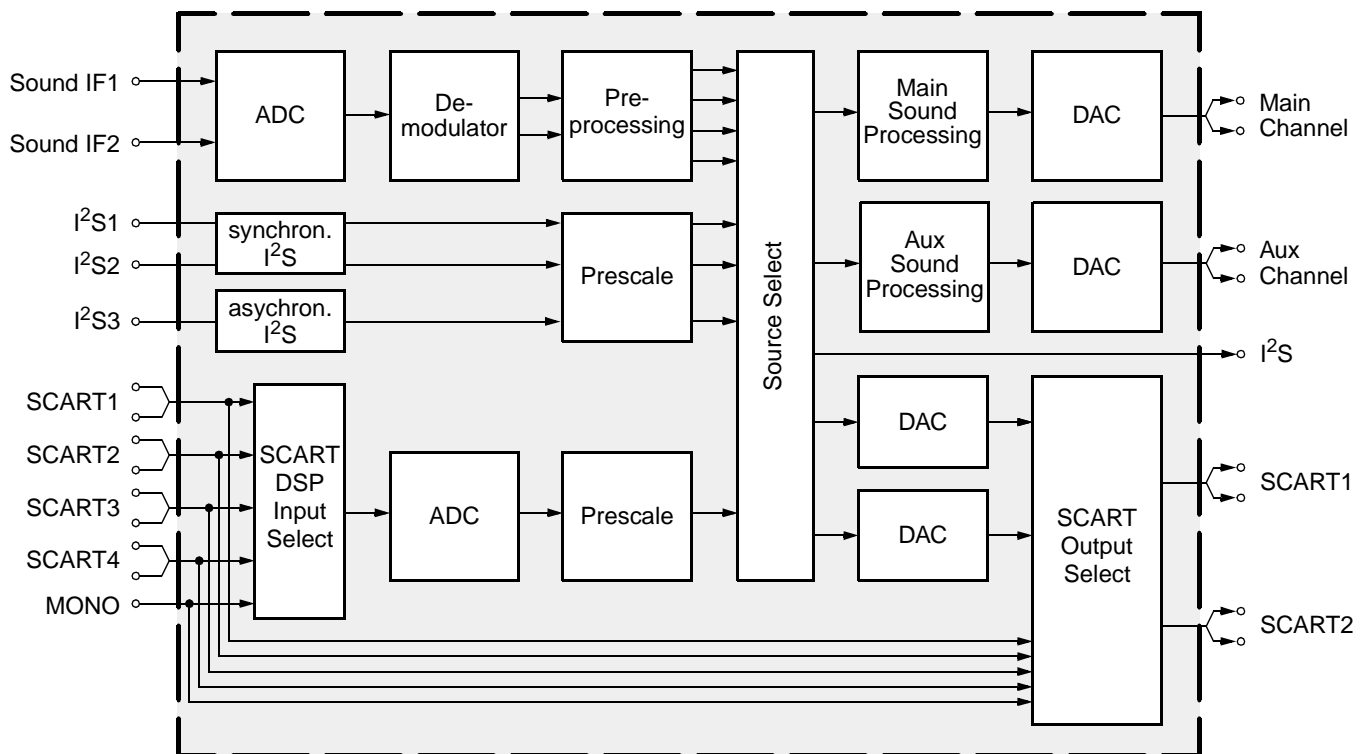


Fig. 1-1: Simplified functional block diagram of the MSP 44x8G

1.1. Features of the MSP 44x8G Family

Feature	4408	4418	4428	4448	4458
Standard Selection with single I ² C transmission	X	X	X	X	X
Automatic Standard Detection of terrestrial TV standards	X	X	X	X	X
Automatic Sound Selection (mono/stereo/bilingual), new registers MODUS, STATUS	X	X	X	X	X
Two selectable sound IF (SIF) inputs	X	X	X	X	X
Automatic Carrier Mute function	X	X	X	X	X
Interrupt output programmable (indicating status change)	X	X	X	X	X
Main/Aux channel with volume, balance, bass, treble, loudness	X	X	X	X	X
AVC: Automatic Volume Correction	X	X	X	X	X
Two channel mixer	X	X	X	X	X
Selectable preemphasis for Aux channel	X	X	X	X	X
Four Stereo SCART (line) inputs, one Mono input; two Stereo SCART outputs	X	X	X	X	X
Complete SCART in/out switching matrix	X	X	X	X	X
Two 48kHz I ² S inputs; one asynchronous 5..50 kHz I ² S input, one 48 kHz I ² S output	X	X	X	X	X
All analog FM-Stereo A2 and satellite standards; AM-SECAM L standard	X	X			X
Simultaneous demodulation of (very) high-deviation FM-Mono and NICAM		X			X
Adaptive deemphasis for satellite (Wegener-Panda, acc. to ASTRA specification)	X	X			X
ASTRA Digital Radio (ADR) together with DRP 3510A	X	X			X
All NICAM standards		X			X
Demodulation of the BTSC multiplex signal and the SAP channel			X	X	X
Alignment free digital DBX noise reduction for BTSC Stereo and SAP				X	X
Alignment free digital Micronas Noise Reduction (MNR) for BTSC Stereo and SAP			X		
BTSC stereo and EIA-J separation significantly better than spec.			X	X	X
SAP and stereo detection for BTSC system			X	X	X
Korean FM-Stereo A2 standard	X	X	X	X	X
Alignment-free Japanese standard EIA-J			X	X	X
Demodulation of the FM-Radio multiplex signal			X	X	X

1.2. MSP 44x8G Version List

Version	Status	Description
MSP 4408G	planned	FM Stereo (A2) Version
MSP 4418G	planned	NICAM and FM Stereo (A2) Version
MSP 4428G	planned	NTSC Version (A2 Korea, BTSC with Micronas Noise Reduction (MNR), and Japanese EIA-J system)
MSP 4448G	planned	NTSC Version (A2 Korea, BTSC with DBX noise reduction, and Japanese EIA-J system)
MSP 4458G	available	Global Version (all sound standards)

1.3. MSP 44x8G Versions and their Application Fields

Table 1–1 provides an overview of TV sound standards that can be processed by the MSP 44x8G family. In addition, the MSP 44x8G is able to handle the terrestrial FM-Radio standard. With the MSP 44x8G, a com-

plete multimedia receiver covering all TV sound standards together with terrestrial and satellite radio sound can be built; even ASTRA Digital Radio can be processed (with a DRP 3510A coprocessor).

Table 1–1: TV Stereo Sound Standards covered by the MSP 44x8G Family (details see Appendix A)

MSP Version		System	Position of Sound Carrier / MHz	Sound Modulation	Color System	Broadcast e.g. in:
4408		B/G	5.5/5.7421875	FM-Stereo (A2)	PAL	Germany
			5.5/5.85	FM-Mono/NICAM	PAL	Scandinavia, Spain
4408		L	6.5/5.85	AM-Mono/NICAM	SECAM-L	France
		I	6.0/6.552	FM-Mono/NICAM	PAL	UK, Hong Kong
		D/K	6.5/5.85	FM-Mono/NICAM	PAL	China, Hungary
			6.5/6.2578125	FM-Stereo (A2, D/K1)	SECAM-East	Slovak. Rep.
6.5/6.7421875	FM-Stereo (A2, D/K2)		PAL	currently no broadcast		
4408		Satellite	6.5	FM-Mono FM-Stereo ASTRA Digital Radio (ADR) with DRP 3510A	PAL	Europe Sat. ASTRA
			7.02/7.2			
			7.38/7.56 etc.			
			4.5/4.724212			
4428/48		M	4.5	FM-FM (EIA-J)	NTSC	Japan
			4.5	BTSC-Stereo + SAP	NTSC	USA
			10.7	FM-Stereo Radio		USA, Europe

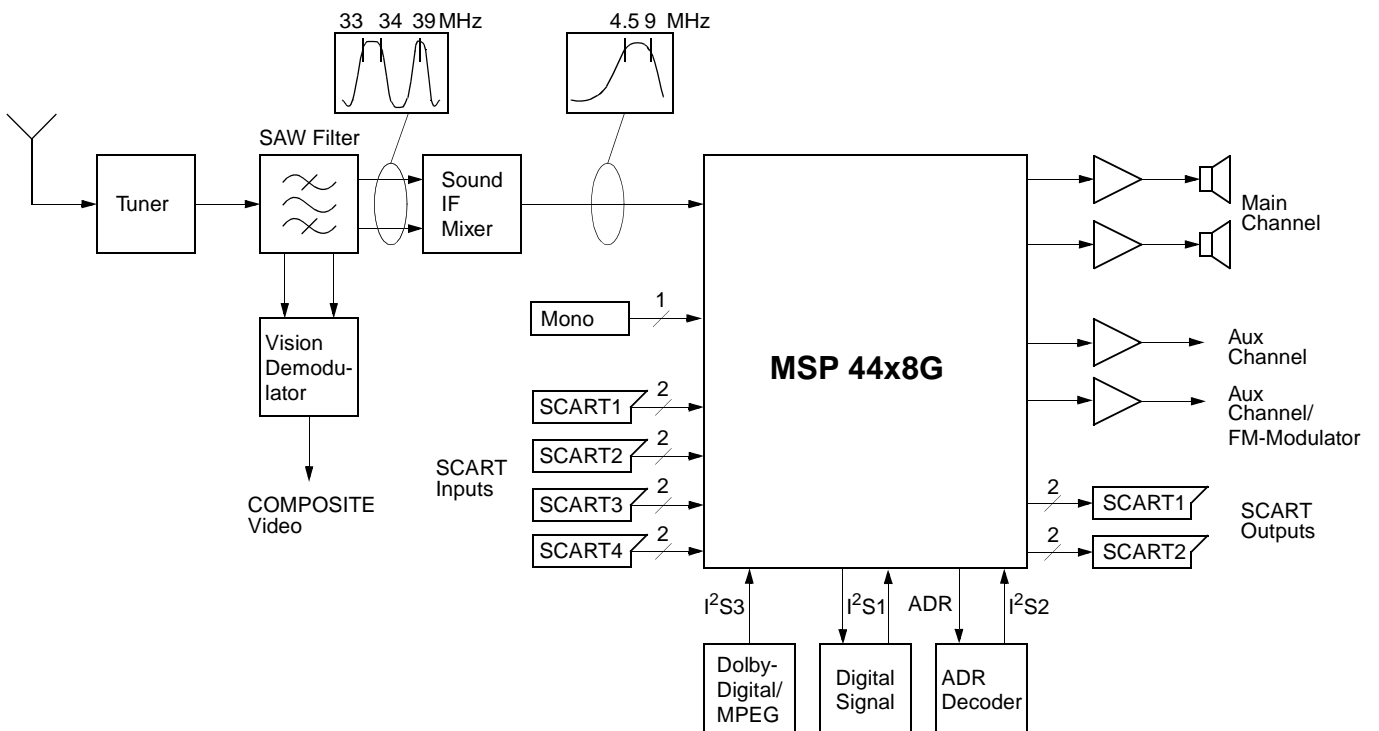


Fig. 1–2: Typical MSP 44x8G application

2. Functional Description

2.1. Architecture of the MSP 44x8G Family

Fig. 2-1 shows a simplified block diagram of the IC. The block diagram contains all features of the

MSP 4458G. Other members of the MSP 44x8G family do not have the complete set of features, handling only a subset of the standards (see dashed block in Fig. 2-1).

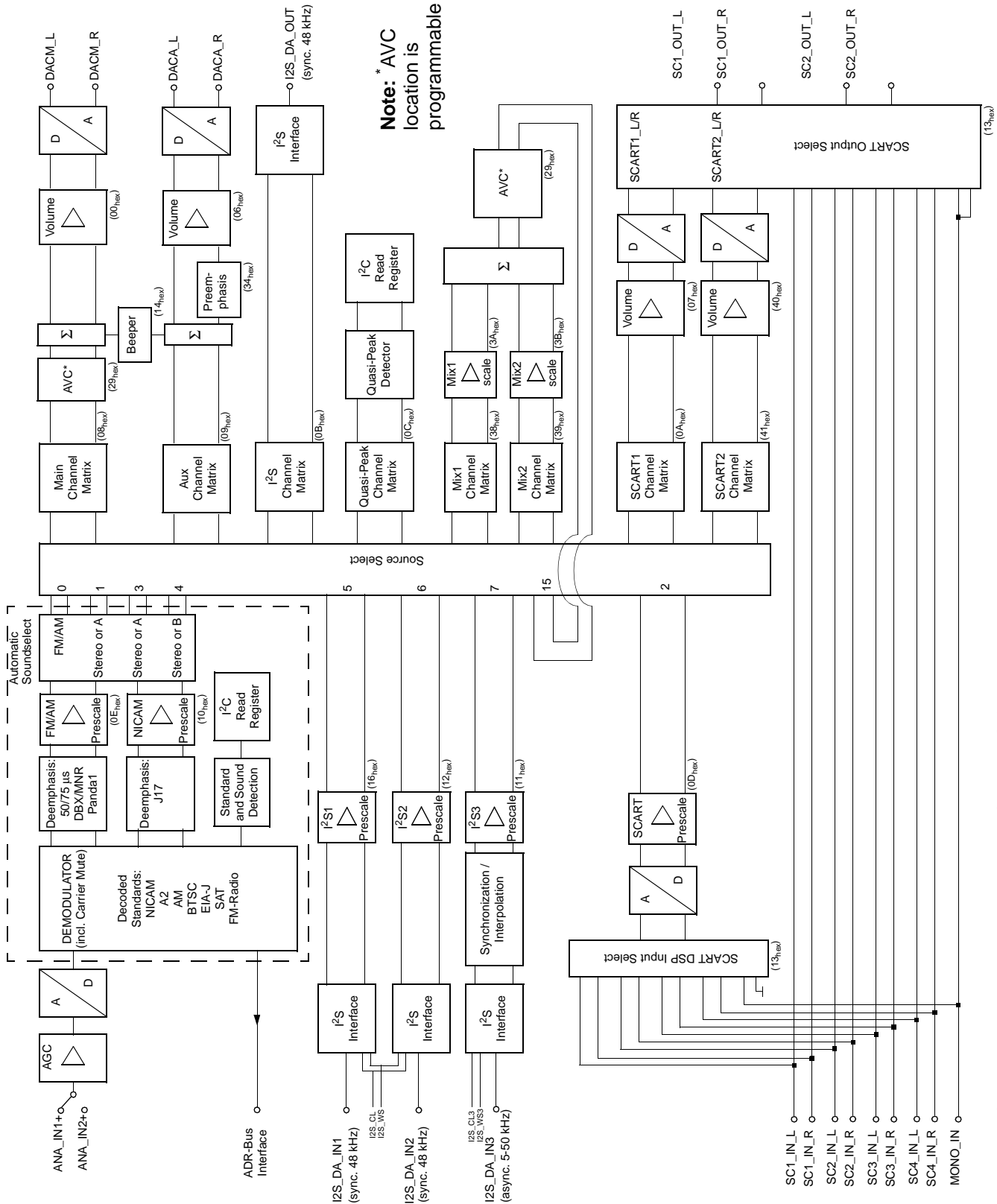


Fig. 2-1: Signal flow block diagram of the MSP 44x8G (input and output names correspond to pin names).

2.2. MSP 44x8G Sound IF Processing

2.2.1. Analog Sound IF Input

The input pins ANA_IN1+, ANA_IN2+, and ANA_IN- offer the possibility to connect two different **sound IF (SIF)** sources to the MSP 44x8G. The preselected sound IF signal is fed into an A/D-converter. An analog **automatic gain circuit (AGC)** allows a wide range of input levels. The highpass filters, formed by the coupling capacitors at pins ANA_IN1+ and ANA_IN2+ (see Section 7.3. "Application Circuit" on page 83), are sufficient in most cases to suppress video components. Some combinations of SAW filters and sound IF mixer ICs, however, show large picture components on their outputs. In this case, further filtering is recommended.

2.2.2. Demodulator: Standards and Features

The MSP 44x8G is able to demodulate all TV-sound standards worldwide including the digital NICAM system. Depending on the MSP 44x8G version, the following demodulation modes can be performed:

A2 Systems: Detection and demodulation of two separate FM carriers (FM1 and FM2), demodulation and evaluation of the identification signal of carrier FM2.

NICAM Systems: (Only possible in the MSP 4418G and MSP 4458G). Demodulation and decoding of the NICAM carrier, detection and demodulation of the analog FM or AM carrier. For D/K-NICAM, the FM carrier may have a maximum deviation of 384 kHz.

Very high deviation FM-Mono: Detection and robust demodulation of one FM carrier with a maximum deviation of 540 kHz.

BTSC-Stereo: Detection and FM demodulation of the aural carrier resulting in the MTS/MPX signal. Detection and evaluation of the pilot carrier, AM demodulation of the (L-R)-carrier and detection of the SAP subcarrier. Processing of DBX noise reduction or **Micronas Noise Reduction (MNR)**.

BTSC-Mono + SAP: Detection and FM demodulation of the aural carrier resulting in the MTS/MPX signal. Detection and evaluation of the pilot carrier, detection and FM demodulation of the SAP subcarrier. Processing of DBX noise reduction or **Micronas Noise Reduction (MNR)**.

Japan Stereo: Detection and FM demodulation of the aural carrier resulting in the MPX signal. Demodulation and evaluation of the identification signal and FM demodulation of the (L-R)-carrier.

FM-Satellite Sound: Demodulation of one or two FM carriers. Processing of high-deviation mono or narrow bandwidth mono, stereo, or bilingual satellite sound according to the ASTRA specification.

FM-Stereo-Radio: Detection and FM demodulation of the aural carrier resulting in the MPX signal. Detection and evaluation of the pilot carrier and AM demodulation of the (L-R)-carrier.

The demodulator blocks of all MSP 44x8G versions have identical user interfaces. Even completely different systems like the BTSC and NICAM systems are controlled the same way. Standards are selected by means of MSP Standard Codes. Automatic processes handle standard detection and identification without controller interaction. The key features of the MSP 44x8G demodulator blocks are described below.

Standard Selection: The controlling of the demodulator is minimized: All parameters, such as tuning frequencies or filter bandwidth, are adjusted automatically by transmitting one single value to the STANDARD SELECT register. For all standards, specific MSP standard codes are defined.

Automatic Standard Detection: If the TV sound standard is unknown, the MSP 44x8G can automatically detect the actual standard, switch to that standard, and respond the actual MSP standard code.

Automatic Carrier Mute: To prevent noise effects or FM identification problems in the absence of an FM carrier, the MSP 44x8G offers a carrier mute feature, which is activated automatically if the standard is selected by means of the STANDARD SELECT register. If no FM carrier is available at one of the two MSP demodulator channels, the corresponding demodulator output is muted.

2.2.3. Preprocessing of Demodulator Signals

All demodulated signals must be processed by a deemphasis filter and adjusted in level (analog signals must also be dematrixed). The correct deemphasis filters are already selected by setting the standard in the STANDARD SELECT register. The level adjustment has to be done by means of the FM/AM and NICAM prescale registers. The necessary dematrix function depends on the selected sound standard and the actual broadcasted sound mode (mono, stereo, or bilingual). It can be manually set by the FM Matrix Mode register or automatically set by the Automatic Sound Selection.

2.2.4. Automatic Sound Select

In the Automatic Sound Select mode, the dematrix function is automatically selected based on the identification information in the STATUS register. No I²C interaction is necessary when the broadcasted sound mode changes (e.g. from mono to stereo).

The demodulator supports the identification check by switching between mono compatible standards (standards that have the same FM mono carrier) automatically and non-audible. If B/G-FM or B/G-NICAM is selected, the MSP will switch between these standards. The same action is performed for the standards: D/K1-FM, D/K2-FM, and D/K-NICAM. Switching is only done in the absence of any stereo or bilingual identification. If identification is found, the MSP keeps the detected standard.

In case of high bit-error rates, the MSP 44x8G automatically falls back from digital NICAM sound to analog FM or AM mono.

Table 2–1 on page 11 summarizes all actions that take place when Automatic Sound Select is switched on.

To provide more flexibility, the Automatic Sound Select block prepares four different source channels of demodulated sound (Fig. 2–2). By choosing one of the four demodulator channels, the preferred sound mode can be selected by means of the Source Select registers, independent for all MSP-outputs.

The following source channels of demodulated sound are defined:

- **“FM/AM” channel:** Analog mono sound, stereo if available. In case of NICAM, analog mono only (FM or AM mono).
- **“Stereo or A/B” channel:** Analog or digital mono sound, stereo if available. In case of bilingual broadcast, it contains both languages A (left) and B (right).
- **“Stereo or A” channel:** Analog or digital mono sound, stereo if available. In case of bilingual broadcast, it contains language A (on left and right).
- **“Stereo or B” channel:** Analog or digital mono sound, stereo if available. In case of bilingual broadcast, it contains language B (on left and right).

Fig. 2–2 and Table 2–2 show the source channel assignment of the demodulated signals in case of Automatic Sound Select mode for all sound standards (see Section 6.).

Note: The analog primary input channel contains the signal of the mono FM/AM carrier or the L+R signal of the MPX carrier. The secondary input channel contains the signal of the second FM carrier, the L–R signal of the MPX carrier, or the SAP signal.

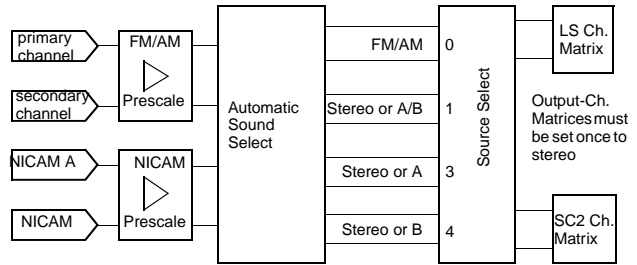


Fig. 2–2: Source channel assignment of demodulated signals in Automatic Sound Select Mode

2.2.5. Manual Mode

Fig. 2–3 shows the source channel assignment of demodulated signals in case of manual mode. If manual mode is required, more information can be found in Section 6.7. “Demodulator Source Channels in Manual Mode” on page 80.

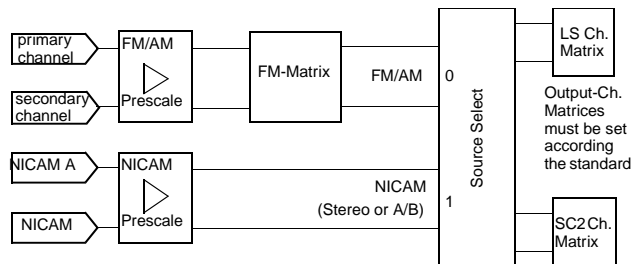


Fig. 2–3: Source channel assignment of demodulated signals in Manual Mode

Table 2–1: Performed actions of the Automatic Sound Selection

Selected TV Sound Standard	Performed Actions
B/G-FM, D/K-FM, M-Korea, and M-Japan	Evaluation of the identification signal and automatic switching to mono, stereo, or bilingual. Preparing four demodulator source channels according to Table 2–2. Identification is acquired after 500 ms.
B/G-NICAM, L-NICAM, I-NICAM, and D/K-NICAM	Evaluation of NICAM-C-bits and automatic switching to mono, stereo, or bilingual. Preparing four demodulator source channels according to Table 2–2. NICAM detection is acquired within 150 ms. In case of bad or no NICAM reception, the MSP switches automatically to FM/AM mono and switches back to NICAM if possible. A hysteresis prevents periodical switching.
B/G-FM, B/G-NICAM or D/K1-FM, D/K2-FM, D/K-NICAM	Automatic searching for stereo/bilingual-identification in case of mono transmission. Automatic and non-audible changes between Dual-FM and FM-NICAM standards while listening to the basic FM-Mono sound carrier. Example: If starting with B/G-FM-Stereo, there will be a periodical alternation to B/G-NICAM in the absence of FM-Stereo/Bilingual or NICAM-identification. Once an identification is detected, the MSP keeps the corresponding standard.
BTSC-STEREO, FM Radio	Evaluation of the pilot signal and automatic switching to mono or stereo. Preparing four demodulator source channels according to Table 2–2. Detection of the SAP carrier. Pilot detection is acquired after 200 ms.
BTSC-SAP	In the absence of SAP, the MSP switches to BTSC-Stereo if available. If SAP is detected, the MSP switches automatically to SAP (see Table 2–2).

Table 2–2: Sound modes for the demodulator source channels with Automatic Sound Select

Broadcasted Sound Standard	Selected MSP Standard Code ³⁾	Broadcasted Sound Mode	Source Channels in Automatic Sound Select Mode			
			FM/AM (source select: 0)	Stereo or A/B (source select: 1)	Stereo or A (source select: 3)	Stereo or B (source select: 4)
M-Korea B/G-FM D/K-FM M-Japan	02 03, 08 ¹⁾ 04, 05, 0B ¹⁾ 30	MONO	Mono	Mono	Mono	Mono
		STEREO	Stereo	Stereo	Stereo	Stereo
		BILINGUAL: Languages A and B	Left = A Right = B	Left = A Right = B	A	B
B/G-NICAM L-NICAM I-NICAM D/K-NICAM D/K-NICAM (with high deviation FM)	08, 03 ²⁾ 09 0A 0B, 04 ²⁾ , 05 ²⁾ 0C	NICAM not available or error rate too high	analog Mono	analog Mono	analog Mono	analog Mono
		MONO	analog Mono	NICAM Mono	NICAM Mono	NICAM Mono
		STEREO	analog Mono	NICAM Stereo	NICAM Stereo	NICAM Stereo
		BILINGUAL: Languages A and B	analog Mono	Left = NICAM A Right = NICAM B	NICAM A	NICAM B
BTSC	20, 21	MONO	Mono	Mono	Mono	Mono
		STEREO	Stereo	Stereo	Stereo	Stereo
	20	MONO+SAP	Mono	Mono	Mono	Mono
		STEREO+SAP	Stereo	Stereo	Stereo	Stereo
	21	MONO+SAP	Left = Mono Right = SAP	Left = Mono Right = SAP	Mono	SAP
		STEREO+SAP	Left = Mono Right = SAP	Left = Mono Right = SAP	Mono	SAP
FM Radio	40	MONO	Mono	Mono	Mono	Mono
		STEREO	Stereo	Stereo	Stereo	Stereo

¹⁾ The Automatic Sound Select process will automatically switch to the mono compatible analog standard.

²⁾ The Automatic Sound Select process will automatically switch to the mono compatible digital standard.

³⁾ The MSP Standard Codes are defined in Table 3–7 on page 20.

2.3. Preprocessing for SCART and I²S Input Signals

The SCART and I²S inputs need only be adjusted in level by means of the SCART and I²S prescale registers.

2.4. Source Selection and Output Channel Matrix

The Source Selector makes it possible to distribute all source signals (one of the demodulator source channels, SCART, or I²S input) to the desired output channels (Main, Aux, etc.). All input and output signals can be processed simultaneously. Each source channel is identified by a unique source address.

For each output channel, the output channel matrix can be set to sound A, sound B, stereo, or mono.

If Automatic Sound Select is on, the output channel matrix can stay fixed to stereo (transparent) for demodulated signals.

2.4.1. Mixing Unit

Any source can be selected as the input for the two channels of the Mixing unit. The mixer channel matrices and the scaling factors can be programmed separately for each channel.

After adding up both channels, the signal is fed back and is available as source 15 (Mix output) of the Source Selector.

2.5. Audio Baseband Processing

2.5.1. Automatic Volume Correction (AVC)

Different sound sources (e.g. terrestrial channels, SAT channels, or SCART) fairly often do not have the same volume level. Advertisements during movies usually have a higher volume level than the movie itself. This results in annoying volume changes. The AVC solves this problem by equalizing the volume level.

In the standard configuration the AVC block is located in the main channel. Alternatively, the AVC function can be moved to the mixer path.

To prevent clipping, the AVC's gain decreases quickly in dynamic boost conditions. To suppress oscillation effects, the gain increases rather slowly for low-level inputs. The decay time is programmable by the AVC register (see page 29).

For input signals ranging from -24 dB_r to 0 dB_r, the AVC maintains a fixed output level of -18 dB_r. Fig. 2-4 shows the AVC output level versus its input level. For prescale and volume registers set to 0 dB, a level of 0 dB_r corresponds to full scale input/output. This is

- SCART input/output 0 dB_r = $2.0 V_{rms}$
- Main and Aux output 0 dB_r = $1.4 V_{rms}$

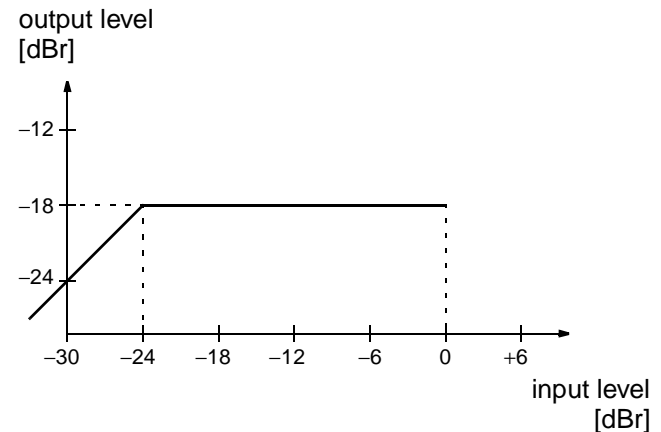


Fig. 2-4: Simplified AVC characteristics

2.5.2. Main and Aux Outputs

The Main and Aux output channels are adjustable in volume. A square wave beeper with adjustable frequency and volume can be added to them.

2.5.3. Quasi-Peak Detector

The Quasi-Peak Readout register can be used to read out the quasi-peak level of any input source. The feature is based on following filter time constants:

- attack time: 1.3 ms
- decay time: 37 ms

2.6. SCART Signal Routing

2.6.1. SCART DSP In and SCART Out Select

The SCART DSP Input Select and SCART Output Select blocks include full matrix switching facilities. To design a TV set with four pairs of SCART-inputs and two pairs of SCART-outputs, no external switching hardware is required. The switches are controlled by the ACB user register (see page 31).

2.6.2. Stand-by Mode

If the MSP 44x8G is switched off by first pulling STANDBYQ low and then (after >1 μ s delay) switching off the 5-V, but keeping the 8-V power supply (**'Stand-by'-mode**), the SCART switches maintain their position and function. This allows the copying from selected SCART-inputs to SCART-outputs in the TV set's stand-by mode.

In case of power on or starting from stand-by (see details on the power-up sequence in Fig. 4–20 on page 56), all internal registers except the ACB register (page 31) are reset to the default configuration (see Table 3–5 on page 18). The reset position of the ACB register becomes active after the first I²C transmission into the Baseband Processing part (subaddress 12_{hex}). By transmitting the ACB register first, the reset state can be redefined.

2.7. I²S Bus Interfaces

The MSP 44x8G has two kinds of interfaces: synchronous master/slave input/output interfaces running on 48 kHz and an asynchronous slave interface.

The interfaces accept a variety of formats with different sample width, bit-orientation, and wordstrobe timing. All I²S options are set by means of the MODUS or I²S_CONFIG register.

2.7.1. Synchronous I²S-Interface(s)

The synchronous I²S bus interface consists of the pins:

- I2S_DA_IN1, I2S_DA_IN2/3 (I2S_DA_IN2 in PQFP80 package):
I²S serial data input, 16, 18...32 bits per sample.
- I2S_DA_OUT:
I²S serial data output, 16, 18...32 bits per sample.
- I2S_CL:
I²S serial clock.
- I2S_WS:
I²S word strobe signal defines the left and right sample.

If the MSP 44x8G serves as the master on the I²S interface, the clock and word strobe lines are driven by the MSP. In this mode, only 16, 32 bits per sample can be selected. In slave mode, these lines are input to the MSP 44x8G and the MSP clock is synchronized to 384 times the I2S_WS rate (48 kHz). NICAM operation is not possible in slave mode.

An I²S timing diagram is shown in Fig. 4–22 on page 59.

2.7.2. Asynchronous I²S-Interface

The asynchronous I²S slave interface allows the reception of digital stereo signals with arbitrary sample rates from 5 to 50 kHz. The synchronization is performed by means of an adaptive sample rate converter. No oversampling clock is required.

The following pins are used for the asynchronous I²S bus interface:

- I2S_WS3 (serves only as input)
- I2S_CL3 (serves only as input)
- I2S_DA_IN2/3 (I2S_DA_IN3 in PQFP80 package).

The interface accepts I²S-input streams with MSB first and with sample widths of 16,18...32 bits. With left/right alignment and wordstrobe timing polarity, there are additional parameters available for the adaption to a variety of formats in the I²S_CONFIG register (see page 24).

2.8. ADR Bus Interface

For the ASTRA Digital Radio System (ADR), the MSP 4408G, MSP 4418G, and MSP 4458G performs preprocessing such as carrier selection and filtering. Via the 3-line ADR-bus, the resulting signals are transferred to the DRP 3510A coprocessor, where the source decoding is performed. To be prepared for an upgrade to ADR with an additional DRP board, the following lines of MSP 44x8G should be provided on a feature connector:

- AUD_CL_OUT
- I2S_DA_IN1, 2, or 3
- I2S_DA_OUT, I2S_WS, I2S_CL
- ADR_CL, ADR_WS, ADR_DA

For more details, please refer to the DRP 3510A data sheet.

2.9. Digital Control I/O Pins and Status Change Indication

The static level of the digital input/output pins D_CTR_I/O_0/1 is switchable between HIGH and LOW via the I²C-bus by means of the ACB register (see page 31). This enables the controlling of external hardware switches or other devices via I²C-bus.

The digital input/output pins can be set to high impedance by means of the MODUS register (see page 23). In this mode, the pins can be used as input. The current state can be read out of the STATUS register (see page 25).

Optionally, the pin D_CTR_I/O_1 can be used as an interrupt request signal to the controller, indicating any changes in the read register STATUS. This makes polling unnecessary, I²C bus interactions are reduced to a minimum (see "STATUS Register" on page 25 and "MODUS Register" on page 23).

2.10. Preemphasis

When using the Aux output for feeding an external modulator, a preemphasis can be applied to the right channel.

The signal is scaled down by –3 dB. An overmodulation protection is included in the algorithm which limits the output signal to 0 dBFS. Due to the nature of a preemphasis, its gain at high frequencies exceeds 3 dB. Thus, even with 0 dB input signals and prescaler / volume set to 0 dB, clipping can occur.

There are three modes present: preemphasis off, 50 μs, and 75 μs. (see Table 3–11 on page 29) for the register settings.

2.11. Clock PLL Oscillator and Crystal Specifications

The MSP 44x8G derives all internal system clocks from the 18.432 MHz oscillator. In NICAM or in I²S-Slave mode of the synchronous interface, the clock is phase-locked to the corresponding source. Therefore, it is not possible to use NICAM and I²S-Slave mode of the synchronous interface at the same time.

For proper performance, the MSP clock oscillator requires a 18.432-MHz crystal. Note that for the phase-locked modes (NICAM, I²S-Slave), crystals with tighter tolerance are required. Please note also, that the asynchronous I²S3 slave interface uses a different locking mechanism and does not require tighter crystal tolerances.

Remark on using the crystal:

External capacitors at each crystal pin to ground are required. They are necessary for tuning the open-loop frequency of the internal PLL and for stabilizing the frequency in closed-loop operation. The higher the capacitors, the lower the resulting clock frequency. The nominal free running frequency should match 18.432 MHz as closely as possible.

Clock measurements should be done at pin AUD_CL_OUT. This pin must be activated for this purpose (see MODUS register on page 23).

3. Control Interface

3.1. I²C Bus Interface

3.1.1. Device and Subaddresses

The MSP 44x8G is controlled via the I²C bus slave interface.

The IC is selected by transmitting one of the MSP 44x8G device addresses. In order to allow up to three MSP ICs to be connected to a single bus, an address select pin (ADR_SEL) has been implemented. With ADR_SEL pulled to high, low, or left open, the MSP 44x8G responds to different device addresses. A device address pair is defined as a write address and a read address (see Table 3–1).

Writing is done by sending the device write address, followed by the subaddress byte, two address bytes, and two data bytes. Reading is done by sending the write device address, followed by the subaddress byte and two address bytes. Without sending a stop condition, reading of the addressed data is completed by sending the device read address and reading two bytes of data. Refer to Section 3.1.2. for the I²C bus protocol and to Section 3.4. “Programming Tips” on page 34 for proposals of MSP 44x8G I²C telegrams. See Table 3–2 for a list of available subaddresses.

Besides the possibility of hardware reset, the MSP can also be reset by means of the RESET bit in the CONTROL register by the controller via I²C bus.

Due to the internal architecture of the MSP 44x8G, the IC cannot react immediately to an I²C request. The typical response time is about 0.3 ms. If the MSP cannot accept another complete byte of data until it has performed some other function (for example, servicing an internal interrupt), it will hold the clock line I2C_CL low to force the transmitter into a wait state. The positions within a transmission where this may happen are indicated by “Wait” in Section 3.1.3. The maximum wait period of the MSP during normal operation mode is less than 1 ms.

Internal hardware error handling:

In case of any internal hardware error (e.g. interruption of the power supply of the MSP), the MSP’s wait period is extended to 1.8 ms. After this time period elapses, the MSP releases data and clock lines.

Indication and solving of the error status:

To indicate the error status, the remaining acknowledge bits of the actual I²C-protocol will be left high. Additionally, bit[14] of CONTROL is set to one. The MSP can then be reset via the I²C bus by transmitting the reset condition to CONTROL.

Indication of reset:

Any reset, even caused by an unstable reset line etc., is indicated in bit[15] of CONTROL.

A general timing diagram of the I²C Bus is shown in Fig. 4–21 on page 57.

Table 3–1: I²C Bus Device Addresses

ADR_SEL	Low		High		Left Open	
	Write	Read	Write	Read	Write	Read
MSP device address	80 _{hex}	81 _{hex}	84 _{hex}	85 _{hex}	88 _{hex}	89 _{hex}

Table 3–2: I²C Bus Subaddresses

Name	Binary Value	Hex Value	Mode	Function
CONTROL	0000 0000	00	Read/Write	Write: Software reset of MSP (see Table 3–3) Read: Hardware error status of MSP
TEST	0000 0001	01	Write	only for internal use
WR_DEM	0001 0000	10	Write	write address demodulator
RD_DEM	0001 0001	11	Write	read address demodulator
WR_DSP	0001 0010	12	Write	write address DSP
RD_DSP	0001 0011	13	Write	read address DSP

3.1.2. Description of CONTROL Register

Table 3–3: CONTROL as a Write Register

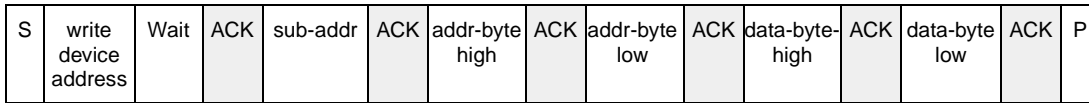
Name	Subaddress	Bit[15] (MSB)	Bits[14:0]
CONTROL	00 _{hex}	1 : RESET 0 : normal	0

Table 3–4: CONTROL as a Read Register

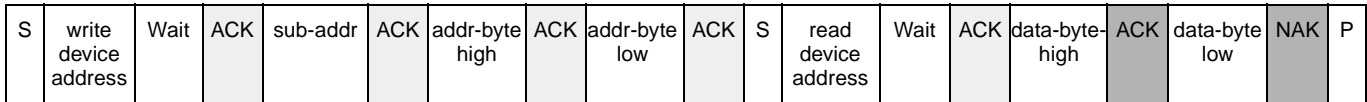
Name	Subaddress	Bit[15] (MSB)	Bit[14]	Bits[13:0]
CONTROL	00 _{hex}	Reset status after last reading of CONTROL: 0 : no reset occurred 1 : reset occurred	Internal hardware status: 0 : no error occurred 1 : internal error occurred	not of interest
Reading of CONTROL will reset the bits[15,14] of CONTROL. After Power-on, bit[15] of CONTROL will be set; it must be read once to be reset.				

3.1.3. Protocol Description

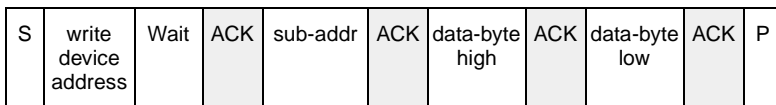
Write to DSP or Demodulator



Read from DSP or Demodulator



Write to Control or Test Registers



Note: S = I²C-Bus Start Condition from master
 P = I²C-Bus Stop Condition from master
 ACK = Acknowledge-Bit: LOW on I2C_DA from slave (= MSP, light gray) or master (= controller, dark gray)
 NAK = Not Acknowledge-Bit: HIGH on I2C_DA from master (dark gray) to indicate 'End of Read' or from MSP indicating internal error state
 Wait = I²C-Clock line is held low, while the MSP is processing the I²C command.

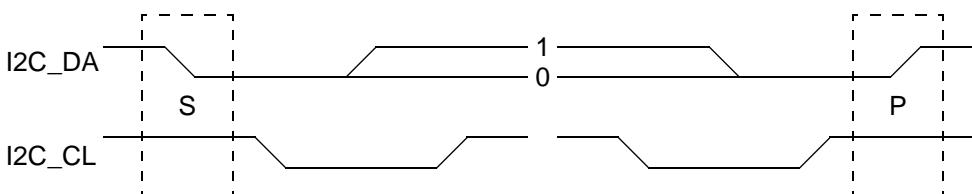


Fig. 3–1: I²C bus protocol (MSB first; data must be stable while clock is high)

3.1.4. Proposals for General MSP 44x8G I²C Telegrams

3.1.4.1. Symbols

daw	write device address (80 _{hex} , 84 _{hex} or 88 _{hex})
dar	read device address (81 _{hex} , 85 _{hex} or 89 _{hex})
<	Start Condition
>	Stop Condition
aa	Address Byte
dd	Data Byte

3.1.4.2. Write Telegrams

<daw 00 d0 00>	write to CONTROL register
<daw 10 aa aa dd dd>	write data into demodulator
<daw 12 aa aa dd dd>	write data into DSP

3.1.4.3. Read Telegrams

<daw 11 aa aa <dar dd dd>	read data from demodulator
<daw 13 aa aa <dar dd dd>	read data from DSP

3.1.4.4. Examples

<80 00 80 00>	RESET MSP statically
<80 00 00 00>	Clear RESET
<80 10 00 20 00 03>	Set demodulator to stand. 03 _{hex}
<80 11 02 00 <81 dd dd>	Read STATUS
<80 12 00 08 01 20>	Set main channel source to NICAM and Matrix to STEREO

More examples of typical application protocols are listed in Section 3.4. "Programming Tips" on page 34.

3.2. Start-Up Sequence: Power-Up and I²C Controlling

After POWER ON or RESET (see Fig. 4–20 on page 56), the IC is in an inactive state. All registers are in the reset position (see Table 3–5 and Table 3–6), the analog outputs are muted. The controller has to initialize all registers for which a non-default setting is necessary.

3.3. MSP 44x8G Programming Interface

3.3.1. User Registers Overview

The MSP 44x8G is controlled by means of user registers. The complete list of all user registers is given in the following tables. The registers are partitioned into the demodulator section (subaddress 10_{hex} for writing, 11_{hex} for reading) and the baseband processing sections (subaddress 12_{hex} for writing, 13_{hex} for reading).

Write and read registers are 16-bit wide, whereby the MSB is denoted bit[15]. Transmissions via I²C bus have to take place in 16-bit words (two byte transfers, with the most significant byte transferred first). All write registers, except the demodulator write registers, are readable.

Unused parts of the 16-bit write registers must be zero. **Addresses not given in this table must not be written.**

An overview of all MSP 44x8G write registers is shown in Table 3–5; all read registers are given in Table 3–6.

Additional read and write registers, together with a detailed description of the manual mode, can be found in the "Appendix B: Manual Mode" on page 73.

Table 3–5: List of MSP 44x8G Write Registers

Write Register	Address (hex)	Bits	Description and Adjustable Range	Reset	See Page
I²C Subaddress = 10_{hex} ; Registers are <i>not</i> readable					
STANDARD SELECT	00 20	[15:0]	Initial Programming of complete Demodulator	00 00	21
MODUS	00 30	[15:0]	Demodulator, Automatic and I ² S options	00 00	22
I²C Subaddress = 12_{hex} ; Registers are <i>all</i> readable by using I²C Subaddress = 13_{hex}					
Volume main channel	00 00	[15:8]	[+12 dB ... -114 dB, MUTE]	MUTE	29
		[7:5] [4:0]	1/8 dB Steps must be set to 0	000 _{bin} 00000 _{bin}	
Volume Aux channel	00 06	[15:8]	[+12 dB ... -114 dB, MUTE]	MUTE	29
		[7:5] [4:0]	1/8 dB Steps must be set to 0	000 _{bin} 00000 _{bin}	
Volume SCART1 output channel	00 07	[15:8]	[+12 dB ... -114 dB, MUTE]	MUTE	30
Main source select	00 08	[15:8]	[FM/AM, NICAM, SCART, I ² S1..3, Mix output]	FM/AM	28
Main channel matrix		[7:0]	[SOUNDA, SOUNDB, STEREO, MONO]	SOUNDA	28
Aux source select	00 09	[15:8]	[FM/AM, NICAM, SCART, I ² S1..3, Mix output]	FM/AM	28
Aux channel matrix		[7:0]	[SOUNDA, SOUNDB, STEREO, MONO]	SOUNDA	28
SCART1 source select	00 0A	[15:8]	[FM/AM, NICAM, SCART, I ² S1..3, Mix output]	FM/AM	28
SCART1 channel matrix		[7:0]	[SOUNDA, SOUNDB, STEREO, MONO]	SOUNDA	28
I ² S source select	00 0B	[15:8]	[FM/AM, NICAM, SCART, I ² S1..3, Mix output]	FM/AM	28
I ² S channel matrix		[7:0]	[SOUNDA, SOUNDB, STEREO, MONO]	SOUNDA	28
Quasi-peak detector source select	00 0C	[15:8]	[FM/AM, NICAM, SCART, I ² S1..3, Mix output]	FM/AM	28
Quasi-peak detector matrix		[7:0]	[SOUNDA, SOUNDB, STEREO, MONO]	SOUNDA	28
Prescale SCART input	00 0D	[15:8]	[00 _{hex} ... 7F _{hex}]	00 _{hex}	27
Prescale FM/AM	00 0E	[15:8]	[00 _{hex} ... 7F _{hex}]	00 _{hex}	26
FM matrix		[7:0]	[NO_MAT, GSTEREO, KSTEREO]	NO_MAT	27
Prescale NICAM	00 10	[15:8]	[00 _{hex} ... 7F _{hex}]	00 _{hex}	27
Prescale I ² S3	00 11	[15:8]	[00 _{hex} ... 7F _{hex}]	10 _{hex}	27
Prescale I ² S2	00 12	[15:8]	[00 _{hex} ... 7F _{hex}]	10 _{hex}	27
SCART switches and D_CTR_I/O	00 13	[15:0]	Bits [15:0]	00 _{hex}	31
Beeper	00 14	[15:0]	[00 _{hex} ... 7F _{hex}]/[00 _{hex} ... 7F _{hex}]	00/00 _{hex}	32
Prescale I ² S1	00 16	[15:8]	[00 _{hex} ... 7F _{hex}]	10 _{hex}	27
AVC: Automatic Volume Correction	00 29	[15:8]	[off, on, decay time]	off	29
Aux Preemphasis on right channel	00 34	[15:8]	[OFF, 50µs, 75µs]	OFF	29
Mix1 source select	00 38	[15:8]	[FM/AM, NICAM, SCART, I ² S1..3, Mix output]	FM/AM	28
Mix1 channel matrix		[7:0]	[SOUNDA, SOUNDB, STEREO, MONO]	SOUNDA	28
Mix2 source select	00 39	[15:8]	[FM/AM, NICAM, SCART, I ² S1..3, Mix output]	FM/AM	28
Mix2 channel matrix		[7:0]	[SOUNDA, SOUNDB, STEREO, MONO]	SOUNDA	28
Scale Mix1	00 3A	[15:8]	[00 _{hex} ... 7F _{hex}]	00 _{hex}	32
Scale Mix2	00 3B	[15:8]	[00 _{hex} ... 7F _{hex}]	00 _{hex}	32

Table 3–5: List of MSP 44x8G Write Registers, continued

Write Register	Address (hex)	Bits	Description and Adjustable Range	Reset	See Page
Volume SCART2 output channel	00 40	[15:8]	[+12 dB ... -114 dB, MUTE]	00 _{hex}	30
SCART2 source select	00 41	[15:8]	[FM/AM, NICAM, SCART, I ² S1..3, Mix output]	FM	28
SCART2 channel matrix		[7:0]	[SOUNDA, SOUNDB, STEREO, MONO]	SOUNDA	28

Table 3–6: List of MSP 44x8G Read Registers

Read Register	Address (hex)	Bits	Description and Adjustable Range	See Page
I²C Subaddress = 11_{hex} : Registers are <i>not</i> writable				
STANDARD RESULT	00 7E	[15:0]	Result of Automatic Standard Detection (see Table 3–8)	25
STATUS	02 00	[15:0]	Monitoring of settings e.g. Stereo, Mono, Mute, D_CTR_I/O etc. .	25
I²C Subaddress = 13_{hex} : Registers are <i>not</i> writable				
Quasi peak readout left	00 19	[15:0]	[00 _{hex} ... 7FFF _{hex}]16 bit two's complement	33
Quasi peak readout right	00 1A	[15:0]	[00 _{hex} ... 7FFF _{hex}]16 bit two's complement	33
MSP hardware version code	00 1E	[15:8]	[00 _{hex} ... FF _{hex}]	33
MSP major revision code		[7:0]	[00 _{hex} ... FF _{hex}]	33
MSP product code	00 1F	[15:8]	[00 _{hex} ... FF _{hex}]	33
MSP ROM version code		[7:0]	[00 _{hex} ... FF _{hex}]	33

3.3.2. Description of User Registers

Table 3–7: Standard Codes for STANDARD SELECT register

MSP Standard Code (Data in hex)	TV Sound Standard	Sound Carrier Frequencies in MHz	MSP 44x8G Version
Automatic Standard Detection			
00 01	Start Automatic Standard Detection		all
Standard Selection			
00 02	M-Dual FM-Stereo	4.5/4.724212	4408, 4418, 4448, 4458
00 03	B/G-Dual FM-Stereo ¹⁾	5.5/5.7421875	4408, 4418, 4458
00 04	D/K1-Dual FM-Stereo ²⁾	6.5/6.2578125	
00 05	D/K2-Dual FM-Stereo ²⁾	6.5/6.7421875	
00 06	D/K-FM-Mono with HDEV3 ³⁾ , not detectable by Automatic Standard Detection, for China HDEV3 ³⁾ SAT-Mono (i.e. Eutelsat, see Table 6–12)	6.5	
00 07	D/K3-Dual FM-Stereo	6.5/5.7421875	4408, 4418, 4458
00 08	B/G-NICAM-FM ¹⁾	5.5/5.85	4418, 4458
00 09	L-NICAM-AM	6.5/5.85	
00 0A	I-NICAM-FM	6.0/6.552	
00 0B	D/K-NICAM-FM ²⁾	6.5/5.85	
00 0C	D/K-NICAM-FM with HDEV2 ⁴⁾ , not detectable by Automatic Standard Detection, for China	6.5/5.85	
00 0D	D/K-NICAM-FM with HDEV3 ³⁾ , not detectable by Automatic Standard Detection, for China	6.5/5.85	4418, 4458
00 20	BTSC-Stereo	4.5	4438, 4448, 4458
00 21	BTSC-Mono + SAP		
00 30	EIA-J Japan Stereo	4.5	4448, 4458
00 40	FM-Stereo Radio	10.7	4438, 4448, 4458
00 50	SAT-Mono (see Table 6–12)	6.5	4408, 4418, 4458
00 51	SAT-Stereo (see Table 6–12)	7.02/7.20	4408, 4418, 4458
00 60	SAT ADR (Astra Digital Radio)	6.12	4408, 4418, 4458
¹⁾ In case of Automatic Sound Select, the B/G-codes 3 _{hex} and 8 _{hex} are equivalent. ²⁾ In case of Automatic Sound Select, the D/K-codes 4 _{hex} , 5 _{hex} , 7 _{hex} and B _{hex} are equivalent. ³⁾ HDEV3: Max. FM deviation must not exceed 540 kHz ⁴⁾ HDEV2: Max. FM deviation must not exceed 360 kHz			

3.3.2.1. STANDARD SELECT Register

The TV sound standard of the MSP 44x8G demodulator is determined by the STANDARD SELECT register. There are two ways to use the STANDARD SELECT register:

- Setting up the demodulator for a TV sound standard by sending the corresponding standard code with a single I²C-Bus transmission.
- Starting the Automatic Standard Detection for terrestrial TV standards. This is the most comfortable way to set up the demodulator. Within 0.5 s, the detection and set-up of the actual TV sound standard is performed. The detected standard can be read out of the STANDARD RESULT register by the control processor. This feature is recommended for the primary set-up of a TV set. Outputs should be muted during Automatic Standard Detection.

The Standard Codes are listed in Table 3–7.

Selecting a TV sound standard via the STANDARD SELECT register initializes the demodulator. This includes: AGC, tuning frequency, band-pass filters, demodulation mode (FM, AM, or NICAM), carrier mute, deemphasis, and identification mode.

If a present sound standard is impossible for a specific MSP version, it switches to the analog mono sound of this standard. In that case, stereo or bilingual processing will not be possible.

For a complete setup of the TV sound processing from analog IF input to the source selection, the following transmissions are necessary: MODUS register, STANDARD SELECT register, prescale values, FM matrix.

Note: The FM matrix is set automatically if Automatic Sound Select is active (MODUS[0]=1). In this case, the FM matrix will be initialized with “Sound A Mono”. During operation, the FM matrix will be automatically selected according to the actual identification information.

3.3.2.2. STANDARD RESULT Register

If Automatic Standard Detection is selected in the STANDARD SELECT register, status and result of the Automatic Standard Detection process can be read out of the STANDARD RESULT register. The possible results are based on the mentioned Standard Code and are listed in Table 3–8.

In cases where no sound standard has been detected (no standard present, too much noise, strong interferers, etc.) the STANDARD RESULT register contains 00 00_{hex}. In that case, the controller has to start further actions (for example, set the standard according to a preference list or by manual input).

As long as the STANDARD RESULT register contains a value greater than 07 FF_{hex}, the Automatic Standard Detection is still active. During this period, the MODUS and STANDARD SELECT register must not be written. The STATUS register will be updated when the Automatic Standard Detection has finished.

If a present sound standard is impossible for a specific MSP version, it detects and switches to the analog mono sound of this standard.

Example:

The MSPs 4438G and 4448G will detect a B/G-NICAM signal as standard 3 and will switch to the analog FM-Mono sound.

Table 3–8: Results of the Automatic Standard Detection

Broadcasted Sound Standard	STANDARD RESULT Register Read 007E _{hex}
Automatic Standard Detection could not find a sound standard	0000 _{hex}
B/G-FM	0003 _{hex}
B/G-NICAM	0008 _{hex}
I	000A _{hex}
FM-Radio	0040 _{hex}
M-FM EIA-J BTSC	0002 _{hex} (if MODUS[14,13]=00)
	0020 _{hex} (if MODUS[14,13]=01)
	0030 _{hex} (if MODUS[14,13]=10)
L-AM D/K1 D/K2	0009 _{hex} (if MODUS[12]=0)
	0004 _{hex} (if MODUS[12]=1)
L-NICAM D/K-NICAM	0009 _{hex} (if MODUS[12]=0)
	000B _{hex} (if MODUS[12]=1)
Automatic Standard Detection still active	>07FF _{hex}

3.3.2.3. Write Registers on I²C Subaddress 10_{hex}

Table 3–9: Write Registers on I²C Subaddress 10_{hex}

Register Address	Function	Name
STANDARD SELECTION		
00 20 _{hex}	STANDARD SELECTION Register Defines TV Sound or FM-Radio Standard bit[15:0] 00 01 _{hex} start Automatic Standard Detection 00 02 _{hex} Standard Codes (see Table 3–7)) ... 00 60 _{hex}	STANDARD_SEL

Table 3–9: Write Registers on I²C Subaddress 10_{hex}, continued

Register Address	Function	Name
MODUS		
00 30 _{hex}	<p>MODUS Register</p> <p>General MSP 44x8G Options</p> <p>bit[15] 0 undefined, must be 0</p> <p>bit[14:13] detected 4.5 MHz carrier is interpreted as:¹⁾</p> <p> 0 standard M (Korea)</p> <p> 1 standard M (BTSC)</p> <p> 2 standard M (Japan)</p> <p> 3 Carrier at 4.5 MHz is ignored (chroma carrier)</p> <p>Preference in Automatic Standard Detection:</p> <p>bit[12] detected 6.5 MHz carrier is interpreted as:¹⁾</p> <p> 0 standard L (SECAM)</p> <p> 1 standard D/K1, D/K2, or D/K NICAM</p> <p>bit[11:9] 0 undefined, must be 0</p> <p>bit[8] 0/1 ANA_IN_1+/ANA_IN_2+; select analog sound IF input pin</p> <p>bit[7] 0/1 active/tristate state of audio clock output pin AUD_CL_OUT</p> <p>bit[6] word strobe alignment (synchronous I²S)</p> <p> 0 WS changes at data word boundary</p> <p> 1 WS changes one clock cycle in advance</p> <p>bit[5] 0/1 master/slave mode of I²S interface (must be set to 0 (= Master) in case of NICAM mode)</p> <p>bit[4] 0/1 active/tristate state of I²S output pins</p> <p>bit[3] state of digital output pins D_CTR_I/O_0 and _1</p> <p> 0 active: D_CTR_I/O_0 and _1 are output pins (can be set by means of the ACB register. see also: MODUS[1])</p> <p> 1 tristate: D_CTR_I/O_0 and _1 are input pins (level can be read out of STATUS[4,3])</p> <p>bit[2] 0 undefined, must be 0</p> <p>bit[1] 0/1 disable/enable STATUS change indication by means of the digital I/O pin D_CTR_I/O_1 Necessary condition: MODUS[3] = 0 (active)</p> <p>bit[0] 0/1 off/on: Automatic Sound Select</p>	MODUS
¹⁾ Valid at the next start of Automatic Standard Detection.		

Table 3–9: Write Registers on I²C Subaddress 10_{hex}, continued

Register Address	Function	Name
0040 _{hex}	I²S Configuration Register (not mentioned bit combinations must not be used)	I2S_CONFIG
	bit[15:12] 0 undefined, must be set to 0	
	bit[11] 0 I ² S Data alignment (I ² S_3) left aligned	
	1 right aligned	
	bit[10] 1 word strobe polarity (I ² S_3) 0 = right, 1 = left	
	0 1 = right, 0 = left	
	bit[9] 0 word strobe alignment (asynchronous I ² S_3) WS changes at data word boundary	
	1 WS changes one clock cycle in advance	
	bit[8:2] 0 undefined, must be set to 0	
	bit[1:0] 00 I2S_CL frequency and I ² S_DA_OUT sample length 2 * 16Bit (1.536MHz Clk)	
	01 2 * 32Bit (3.072MHz Clk)	
	1x undefined, must not be used	

3.3.2.4. Read Registers on I²C Subaddress 11_{hex}

Table 3–10: Read Registers on I²C Subaddress 11_{hex}

Register Address	Function	Name
STANDARD RESULT		
00 7E _{hex}	<p>STANDARD RESULT Register</p> <p>Readback of the detected TV Sound or FM-Radio Standard</p> <p>bit[15:0] 00 00_{hex} Automatic Standard Detection could not find a sound standard</p> <p> 00 02_{hex} MSP Standard Codes (see Table 3–8)</p> <p> ...</p> <p> 00 40_{hex}</p> <p> >07 FF_{hex} Automatic Standard Detection still active</p>	STANDARD_RES
STATUS		
02 00 _{hex}	<p>STATUS Register</p> <p>Contains all user relevant internal information about the status of the MSP</p> <p>bit[15:10] undefined</p> <p>bit[8] 0/1 “1” indicates bilingual sound mode or SAP present</p> <p>bit[7] 0/1 “1” indicates independent mono sound (only for NICAM on MSP 4418G and MSP 4458G)</p> <p>bit[6] 0/1 mono/stereo indication</p> <p>bit[5,9] 00 analog sound standard (FM or AM) active</p> <p> 01 this pattern will not occur</p> <p> 10 digital sound (NICAM) available (MSP 4418G and MSP 4458G only)</p> <p> 11 bad reception condition of digital sound (NICAM) due to:</p> <p> a. high error rate</p> <p> b. unimplemented sound code</p> <p> c. data transmission only</p> <p>bit[4] 0/1 low/high level of digital I/O pin D_CTR_I/O_1</p> <p>bit[3] 0/1 low/high level of digital I/O pin D_CTR_I/O_0</p> <p>bit[2] 0 detected secondary carrier (2nd A2 or SAP carrier)</p> <p> 1 no secondary carrier detected</p> <p>bit[1] 0 detected primary carrier (Mono or MPX carrier)</p> <p> 1 no primary carrier detected</p> <p>bit[0] undefined</p> <p>If STATUS change indication is activated by means of MODUS[1]: Each change in the STATUS register sets the digital I/O pin D_CTR_I/O_1 to high level. Reading the STATUS register resets D_CTR_I/O_1.</p>	STATUS

3.3.2.5. Write Registers on I²C Subaddress 12_{hex}

Table 3–11: Write Registers on I²C Subaddress 12_{hex}

Register Address	Function	Name
PREPROCESSING		
00 0E _{hex}	<p>FM/AM Prescale</p> <p>bit[15:8] 00_{hex}... 7F_{hex} 00_{hex} Defines the input prescale gain for the demodulated FM or AM signal off (RESET condition)</p> <p>For all FM modes except satellite FM, the combinations of prescale value and FM deviation listed below lead to internal full scale.</p> <p>FM mode</p> <p>bit[15:8] 7F_{hex} 28 kHz FM deviation 48_{hex} 50 kHz FM deviation 30_{hex} 75 kHz FM deviation 24_{hex} 100 kHz FM deviation 18_{hex} 150 kHz FM deviation 13_{hex} 180 kHz FM deviation (limit)</p> <p>FM high deviation mode (HDEV2, MSP Standard Code = C_{hex})</p> <p>bit[15:8] 30_{hex} 150 kHz FM deviation 14_{hex} 360 kHz FM deviation (limit)</p> <p>FM very high deviation mode (HDEV3, MSP Standard Code = 6)</p> <p>bit[15:8] 20_{hex} 450 kHz FM deviation 1A_{hex} 540 kHz FM deviation (limit)</p> <p>Satellite FM with adaptive deemphasis</p> <p>bit[15:8] 10_{hex} recommendation</p> <p>AM mode (MSP Standard Code = 9)</p> <p>bit[15:8] 7C_{hex} recommendation for SIF input levels from 0.1 V_{pp} to 0.8 V_{pp}</p> <p>(Due to the AGC switched on, the AM-output level remains stable and independent of the actual SIF-level in the mentioned input range)</p>	PRE_FM

Table 3–11: Write Registers on I²C Subaddress 12_{hex}, continued

Register Address	Function	Name
(continued) 00 0E _{hex}	<p>FM Matrix Modes</p> <p>Defines the dematrix function for the demodulated FM signal</p> <p>bit[7:0] 00_{hex} no matrix (used for bilingual and unmatrixed stereo sound) 01_{hex} German stereo (Standard B/G) 02_{hex} Korean stereo (also used for BTSC, EIA-J, and FM Radio) 03_{hex} sound A mono (left and right channel contain the mono sound of the FM/AM mono carrier) 04_{hex} sound B mono (i.e. SAP)</p> <p>In case of Automatic Sound Select, the FM Matrix Mode is set automatically, i.e. the low-part of any I²C transmission to the register 00 0E_{hex} is ignored.</p> <p>To enable a Forced Mono Mode for all analog stereo systems by overriding the internal pilot or identification evaluation, the following steps must be transmitted:</p> <ol style="list-style-type: none"> 1. MODUS with bit[0] = 0 (Automatic Sound Select off) 2. FM Presc./Matrix with FM Matrix = Sound A Mono (SAP: Sound B Mono) 3. Select FM/AM source channel, with channel matrix set to “Stereo” (transparent) 	FM_MATRIX
00 10 _{hex}	<p>NICAM Prescale</p> <p>Defines the input prescale value for the digital NICAM signal</p> <p>bit[15:8] 00_{hex} ... 7F_{hex} prescale gain</p> <p>examples:</p> <p>00_{hex} off 20_{hex} 0 dB gain 5A_{hex} 9 dB gain (recommendation) 7F_{hex} +12 dB gain (maximum gain)</p>	PRE_NICAM
00 16 _{hex} 00 12 _{hex} 00 11 _{hex}	<p>I2S1 Prescale I2S2 Prescale I2S3 Prescale</p> <p>Defines the input prescale value for digital I²S input signals</p> <p>bit[15:8] 00_{hex} ... 7F_{hex} prescale gain</p> <p>examples:</p> <p>00_{hex} off 10_{hex} 0 dB gain (recommendation) 7F_{hex} +18 dB gain (maximum gain)</p>	PRE_I2S1 PRE_I2S2 PRE_I2S3
00 0D _{hex}	<p>SCART Input Prescale</p> <p>Defines the input prescale value for the analog SCART input signal</p> <p>bit[15:8] 00_{hex} ... 7F_{hex} prescale gain</p> <p>examples:</p> <p>00_{hex} off 19_{hex} 0 dB gain (2 V_{RMS} input leads to digital full scale) 7F_{hex} +14 dB gain (400 mV_{RMS} input leads to digital full scale)</p>	PRE_SCART

Table 3–11: Write Registers on I²C Subaddress 12_{hex}, continued

Register Address	Function	Name
SOURCE SELECT AND OUTPUT CHANNEL MATRIX		
00 08 _{hex} 00 09 _{hex} 00 0A _{hex} 00 41 _{hex} 00 0B _{hex} 00 0C _{hex} 00 38 _{hex} 00 39 _{hex}	<p>Source for:</p> <p>Main Output Aux Output SCART1 DA Output SCART2 DA Output I²S Output Quasi-Peak Detector Mix1 Input Mix2 Input</p> <p>bit[15:8] 0 “FM/AM”: demodulated FM or AM mono signal 1 “Stereo or A/B”: demodulator Stereo or A/B signal 3 “Stereo or A”: demodulator Stereo Sound or Language A (only defined for Automatic Sound Select) 4 “Stereo or B”: demodulator Stereo Sound or Language B (only defined for Automatic Sound Select) 2 SCART input 5 I²S1 input 6 I²S2 input 7 I²S3 input 15 Mix output</p> <p>For demodulator sources, see Table 2–2.</p>	SRC_MAIN SRC_AUX SRC_SCART1 SRC_SCART2 SRC_I2S SRC_QPEAK SRC_MIX1 SRC_MIX2
00 08 _{hex} 00 09 _{hex} 00 0A _{hex} 00 41 _{hex} 00 0B _{hex} 00 0C _{hex} 00 38 _{hex} 00 39 _{hex}	<p>Matrix Mode for:</p> <p>Main Output Aux Output SCART1 DA Output SCART2 DA Output I²S Output Quasi-Peak Detector Mix1 Input Mix2 Input</p> <p>bit[7:0] 00_{hex} Sound A Mono (or Left Mono) 10_{hex} Sound B Mono (or Right Mono) 20_{hex} Stereo (transparent mode) 30_{hex} Mono (sum of left and right inputs divided by 2) More modes are listed in Section 6.5.1.</p> <p>In Automatic Sound Select mode, the demodulator source channels are set according to Table 2–2. Therefore, the matrix modes of the corresponding output channels should be set to “Stereo” (transparent).</p>	MAT_MAIN MAT_AUX MAT_SCART1 MAT_SCART2 MAT_I2S MAT_QPEAK MAT_MIX1 MAT_MIX2

Table 3–11: Write Registers on I²C Subaddress 12_{hex}, continued

Register Address	Function	Name
MAIN AND AUX PROCESSING		
00 00 _{hex} 00 06 _{hex}	<p>Volume Main Volume Aux</p> <p>bit[15:8] volume table with 1 dB step size 7F_{hex} +12 dB (maximum volume) 7E_{hex} +11 dB ... 74_{hex} +1 dB 73_{hex} 0 dB 72_{hex} -1 dB ... 02_{hex} -113 dB 01_{hex} -114 dB 00_{hex} Mute (reset condition) FF_{hex} Fast Mute (needs about 75 ms until the signal is completely ramped down)</p> <p>bit[7:5] higher resolution volume table 0 +0 dB 1 +0.125 dB increase in addition to the volume table ... 7 +0.875 dB increase in addition to the volume table</p> <p>bit[4:0] not used must be set to 0</p> <p>With large scale input signals, positive volume settings may lead to signal clipping.</p> <p>The MSP 44x8G Main and Aux volume function is divided into a digital and an analog section. With Fast Mute, volume is reduced to mute position by digital volume only. Analog volume is not changed. This reduces any audible DC plops. To turn volume on again, the volume step that has been used before Fast Mute was activated must be transmitted.</p>	VOL_MAIN VOL_AUX
0029 _{hex}	<p>Automatic Volume Correction (AVC)</p> <p>bit[15] 0 AVC off, reset of internal variables 1 AVC on</p> <p>bit[14] 0 AVC in Main path 1 AVC in Mixer path</p> <p>bit[13:12] 0 must be set to zero</p> <p>bit[11:8] 8 8 s decay time 4 4 s decay time (recommended) 2 2 s decay time 1 20 ms decay time (should be used for approx. 100 ms after channel change)</p>	AVC AVC_DECAY
00 34 _{hex}	<p>Preemphasis Aux Channel</p> <p>bit[15:8] 00_{hex} Preemphasis OFF 7F_{hex} Preemphasis 50 μs (-3 dB scaling) FF_{hex} Preemphasis 75 μs (-3 dB scaling)</p>	PREEMP_AUX

Table 3–11: Write Registers on I²C Subaddress 12_{hex}, continued

Register Address	Function	Name
SCART OUTPUT CHANNEL		
00 07 _{hex} 00 40 _{hex}	Volume SCART1 Output Channel Volume SCART2 Output Channel bit[15:8] volume table with 1 dB step size 7F _{hex} +12 dB (maximum volume) 7E _{hex} +11 dB ... 74 _{hex} +1 dB 73 _{hex} 0 dB 72 _{hex} -1 dB ... 02 _{hex} -113 dB 01 _{hex} -114 dB 00 _{hex} Mute (reset condition) bit[7:5] higher resolution volume table 0 +0 dB 1 +0.125 dB increase in addition to the volume table ... 7 +0.875 dB increase in addition to the volume table bit[4:0] 01 _{hex} this must be 01 _{hex}	VOL_SCART1 VOL_SCART2

Table 3–11: Write Registers on I²C Subaddress 12_{hex}, continued

Register Address	Function	Name
SCART SWITCHES AND DIGITAL I/O PINS		
00 13 _{hex}	<p>ACB Register</p> <p>Defines the level of the digital output pins and the position of the SCART switches</p> <p>bit[15] 0/1 low/high of digital output pin D_CTR_I/O_0 (MODUS[3]=0)</p> <p>bit[14] 0/1 low/high of digital output pin D_CTR_I/O_1 (MODUS[3]=0)</p> <p>bit[13:5] SCART DSP Input Select</p> <p> xxxx00 xx0 SCART1 to DSP input (RESET position)</p> <p> xxxx01 xx0 MONO to DSP input (Sound A Mono must be selected in the channel matrix mode for the corresponding output channels)</p> <p> xxxx10 xx0 SCART2 to DSP input</p> <p> xxxx11 xx0 SCART3 to DSP input</p> <p> xxxx00 xx1 SCART4 to DSP input</p> <p> xxxx11 xx1 mute DSP input</p> <p>bit[13:5] SCART1 Output Select</p> <p> xx00xx x0x SCART3 input to SCART1 output (RESET position)</p> <p> xx01xx x0x SCART2 input to SCART1 output</p> <p> xx10xx x0x MONO input to SCART1 output</p> <p> xx11xx x0x SCART1 DA to SCART1 output</p> <p> xx00xx x1x SCART2 DA to SCART1 output</p> <p> xx01xx x1x SCART1 input to SCART1 output</p> <p> xx10xx x1x SCART4 input to SCART1 output</p> <p> xx11xx x1x mute SCART1 output</p> <p>bit[13:5] SCART2 Output Select</p> <p> 00xxxx 0xx SCART1 DA to SCART2 output (RESET position)</p> <p> 01xxxx 0xx SCART1 input to SCART2 output</p> <p> 10xxxx 0xx MONO input to SCART2 output</p> <p> 00xxxx 1xx SCART2 DA to SCART2 output</p> <p> 01xxxx 1xx SCART2 input to SCART2 output</p> <p> 10xxxx 1xx SCART3 input to SCART2 output</p> <p> 11xxxx 1xx SCART4 input to SCART2 output</p> <p> 11xxxx 0xx mute SCART2 output</p> <p>The RESET position becomes active at the time of the first write transmission on the control bus to the audio processing part. By writing to the ACB register first, the RESET state can be redefined.</p>	ACB_REG

Table 3–11: Write Registers on I²C Subaddress 12_{hex}, continued

Register Address	Function	Name
MIXING UNIT		
00 3A _{hex} 00 3B _{hex}	MIX1 Scale MIX2 Scale Defines the input scale value for the digital mixing unit bit[15:8] 00 _{hex} off 20 _{hex} 50% (–6 dB gain) 40 _{hex} 100% (0 dB gain) 7F _{hex} 200% (+6 dB gain = maximum gain) Note: If the sum of both mixing inputs exceeds 100%, clipping may occur in the successive processing.	VOL_MIX1 VOL_MIX2
BEEPER		
00 14 _{hex}	Beeper Volume and Frequency bit[15:8] Beeper Volume 00 _{hex} off 7F _{hex} maximum volume bit[7:0] Beeper Frequency 01 _{hex} 16 Hz (lowest) 40 _{hex} 1 kHz FF _{hex} 4 kHz	BEEPER

3.3.2.6. Read Registers on I²C Subaddress 13_{hex}

Table 3–12: Read Registers on I²C Subaddress 13_{hex}

Register Address	Function	Name
QUASI-PEAK DETECTOR READOUT		
00 19 _{hex} 00 1A _{hex}	<p>Quasi-Peak Detector Readout Left Quasi-Peak Detector Readout Right</p> <p>bit[15:0] 0_{hex}... values are 16 bit two's complement (only positive) 7FFF_{hex}</p>	QPEAK_L QPEAK_R
MSP 44x8G VERSION READOUT Registers		
001E _{hex}	<p>MSP Hardware Version Code bit[15:8] 01_{hex} MSP 44x8G-A1</p> <p>A change in the hardware version code defines hardware optimizations that may have influence on the chip's behavior. The readout of this register is identical to the hardware version code in the chip's imprint.</p> <p>MSP Family Code bit[7:4] 1_{hex} MSP 44x8G-A1</p> <p>MSP Major Revision Code bit[3:0] 7_{hex} MSP 44x8G-A1</p>	MSP_HARD MSP_FAMILY MSP_REVISION
001F _{hex}	<p>MSP Product Code bit[15:8] 08_{hex} MSP 4408G-A1 12_{hex} MSP 4418G-A1 1C_{hex} MSP 4428G-A1 30_{hex} MSP 4448G-A1 3A_{hex} MSP 4458G-A1</p> <p>By means of the MSP-Product Code, the control processor is able to decide which TV sound standards have to be considered.</p> <p>MSP ROM Version Code bit[7:0] 41_{hex} MSP 44x8G-A1</p> <p>A change in the ROM version code defines internal software optimizations, that may have influence on the chip's behavior, e.g. new features may have been included. While a software change is intended to create no compatibility problems, customers that want to use the new functions can identify new MSP 44x8G versions according to this number.</p>	MSP_PRODUCT MSP_ROM

3.4. Programming Tips

This section describes the preferred method for initializing the MSP 44x8G. The initialization is grouped into four sections:

- SCART Signal Path (analog signal path)
- Demodulator Input
- SCART and I²S Inputs
- Output Channels

See Fig. 2–1 on page 8 for a complete signal flow.

SCART Signal Path

1. Select analog input for the SCART baseband processing (SCART DSP Input Select) by means of the ACB register.
2. Select the source for each analog SCART output (SCART Output Select) by means of the ACB register.

Demodulator Input

For a complete setup of the sound processing from analog IF input to the source selection, the following steps must be performed:

1. Set MODUS register to the preferred mode and Sound IF input.
2. Write STANDARD SELECT register.
3. Choose preferred prescale (FM and NICAM) values.
4. If Automatic Sound Select is not active: Choose FM matrix repeatedly according to the sound mode indicated in the STATUS register.

SCART and I²S Inputs

1. Select preferred prescale for SCART.
2. Select preferred prescale for I²S inputs (set to 0 dB after RESET).

Output Channels

1. Select the source channel and matrix for each output channel.
2. Set audio baseband features (i.e. AVC, 75 μ s pre-emphasis)
3. Select volume for each output channel.

3.5. Examples of Minimum Initialization Codes

Initialization of the MSP 44x8G according to these listings reproduces sound of the selected standard on the main output. All numbers are hexadecimal. The examples have the following structure:

1. Perform an I²C controlled reset of the IC.
2. Write MODUS register (with Automatic Sound Select).
3. Set Source Selection for main channel (with matrix set to STEREO).
4. Set Prescale (FM and/or NICAM and dummy FM matrix).
5. Write STANDARD SELECT register.
6. Set Volume main channel to 0 dB.

3.5.1. B/G-FM (A2 or NICAM)

```
<80 00 80 00> // Softreset
<80 00 00 00>
<80 10 00 30 20 03> // MODUS-Register: Automatic = on
<80 12 00 08 03 20> // Source Sel. = (St or A) & Ch. Matr. = St
<80 12 00 0E 24 03> // FM/AM-Prescale = 24hex,
                    // FM-Matrix = MONO/SOUNDA
<80 12 00 10 00 5A> // NICAM-Prescale = 5Ahex
<80 10 00 20 00 03> // Standard Select: A2 B/G or NICAM B/G
                    or
<80 10 00 20 00 08>
<80 12 00 00 73 00> // Main Volume 0 dB
```

3.5.2. BTSC-Stereo

```
<80 00 80 00> // Softreset
<80 00 00 00>
<80 10 00 30 20 03> // MODUS-Register: Automatic = on
<80 12 00 08 03 20> // Source Sel. = (St or A) & Ch. Matr. = St
<80 12 00 0E 24 03> // FM/AM-Prescale = 24hex,
                    // FM-Matrix = Sound A Mono
<80 10 00 20 00 20> // Standard Select: BTSC-STEREO
<80 12 00 00 73 00> // Main Volume 0 dB
```

3.5.3. BTSC-SAP with SAP at Main Channel

```
<80 00 80 00> // Softreset
<80 00 00 00>
<80 10 00 30 20 03> // MODUS-Register: Automatic = on
<80 12 00 08 04 20> // Source Sel. = (St or B) & Ch. Matr. = St
<80 12 00 0E 24 03> // FM/AM-Prescale = 24hex,
                    // FM-Matrix = Sound A Mono
<80 10 00 20 00 21> // Standard Select: BTSC-SAP
<80 12 00 00 73 00> // Main Volume 0 dB
```

3.5.4. FM-Stereo Radio

```

<80 00 80 00> // Softreset
<80 00 00 00>
<80 10 00 30 20 03> // MODUS-Register: Automatic = on
<80 12 00 08 03 20> // Source Sel. = (St or A) & Ch. Matr. = St
<80 12 00 0E 24 03> // FM/AM-Prescale = 24hex,
                    FM-Matrix = Sound A Mono
<80 10 00 20 00 40> // Standard Select: FM-STEREO-RADIO
<80 12 00 00 73 00> // Main Volume 0 dB

```

3.5.5. Automatic Standard Detection

A detailed software flow diagram is shown in Fig. 3–2 on page 36.

```

<80 00 80 00> // Softreset
<80 00 00 00>
<80 10 00 30 20 03> // MODUS-Register: Automatic = on
<80 12 00 08 03 20> // Source Sel. = (St or A) & Ch. Matr. = St
<80 12 00 0E 24 03> // FM/AM-Prescale = 24hex,
                    FM-Matrix = Sound A Mono
<80 12 00 10 00 5A> // NICAM-Prescale = 5Ahex
<80 10 00 20 00 01> // Standard Select:
                    Automatic Standard Detection
// Wait till STANDARD RESULT contains a value ≤ 07FF
// IF STANDARD RESULT contains 0000
// do some error handling
// ELSE
<80 12 00 00 73 00> // Main Volume 0 dB

```

**3.5.6. Software Flow for
Interrupt driven STATUS Check**

A detailed software flow diagram is shown in Fig. 3–2 on page 36.

If the D_CTR_I/O_1 pin of the MSP 44x8G is connected to an interrupt input pin of the controller, the following interrupt handler can be applied to be automatically called with each status change of the MSP 44x8G. The interrupt handler may adjust the display according to the new status information.

Interrupt Handler:

```

<80 11 02 00 <81 dd dd> // Read STATUS
// adjust display with given status information
// Return from Interrupt

```

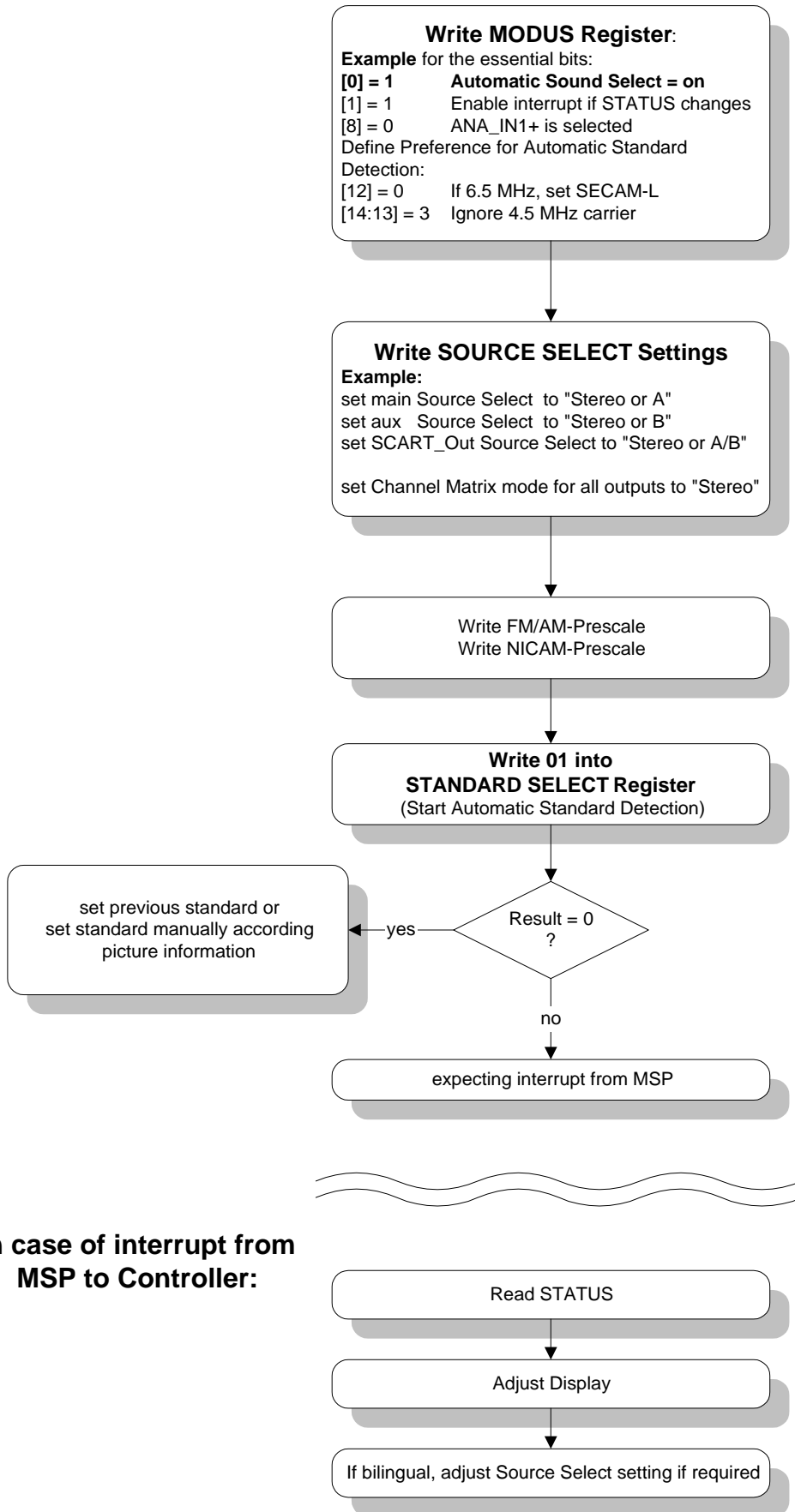
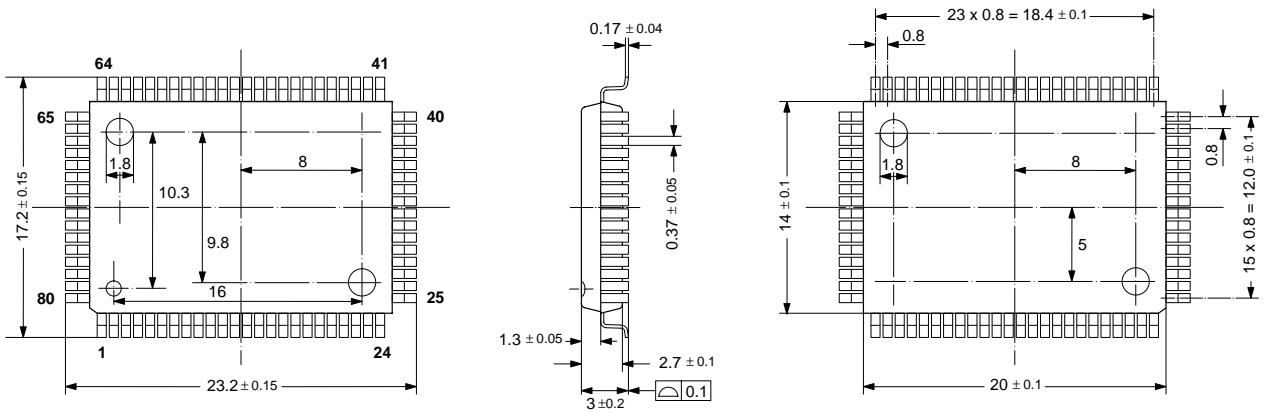


Fig. 3–2: Software flow diagram for a minimum demodulator setup for a European multistandard set applying the Automatic Sound Select feature

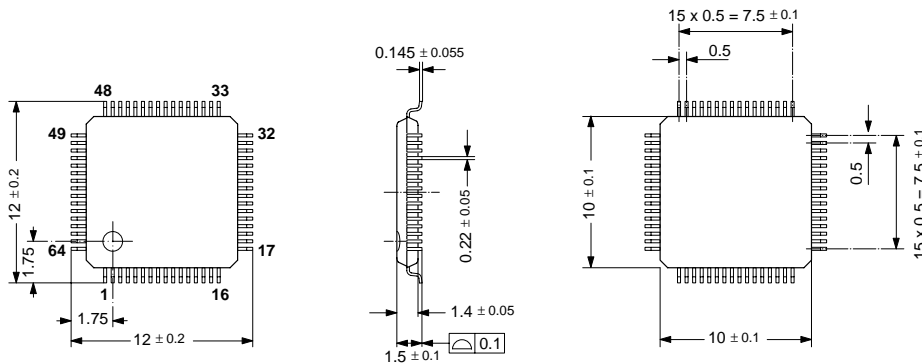
4. Specifications

4.1. Outline Dimensions



SPGS705000-1(P80)/1E

Fig. 4-1:
 80-Pin Plastic Quad Flat Pack
(PQFP80)
 Weight approximately 1.61 g
 Dimensions in mm



D0025/3E

Fig. 4-2:
 64-Pin Plastic Low-Profile Quad Flat Pack
(PLQFP64)
 Weight approximately 0.35 g
 Dimensions in mm

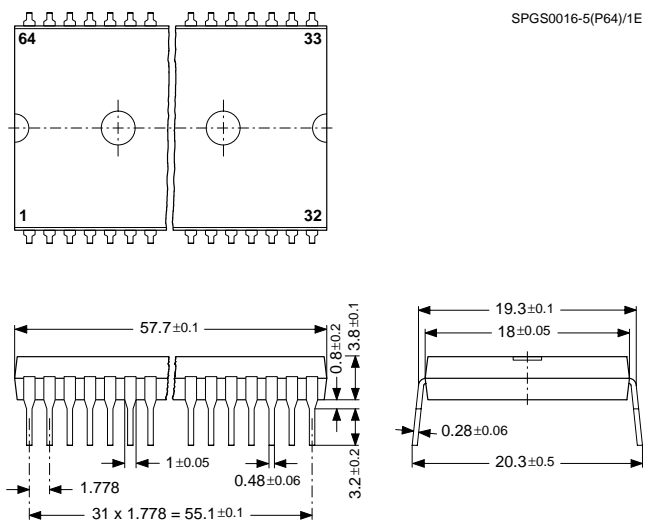


Fig. 4-3:
64-Pin Plastic Shrink Dual-Inline Package
(PSDIP64)
 Weight approximately 9.0 g
 Dimensions in mm

4.2. Pin Connections and Short Descriptions

NC = not connected (**leave vacant** for future compatibility reasons)

TP = Test Pin (**leave vacant** - pin is used for production test only)

LV = leave vacant

X = obligatory; connect as described in application circuit diagram

Pin No.	Pin No.		Pin Name	Type	Connection (if not used)	Short Description
	PQFP 80-pin	PLQFP 64-pin				
1	64	8	NC		LV	Not connected
2	1	9	I2C_CL	IN/OUT	X	I ² C clock
3	2	10	I2C_DA	IN/OUT	X	I ² C data
4	3	11	I2S_CL	IN/OUT	LV	I ² S clock
5	4	12	I2S_WS	IN/OUT	LV	I ² S word strobe
6	5	13	I2S_DA_OUT	OUT	LV	I ² S data output
7	6	14	I2S_DA_IN1	IN	LV	I ² S1 data input
8	7	15	ADR_DA	OUT	LV	ADR data output
9	8	16	ADR_WS	OUT	LV	ADR word strobe
10	9	17	ADR_CL	OUT	LV	ADR clock
11	–	–	DVSUP		X	Digital power supply +5 V
12	–	–	DVSUP		X	Digital power supply +5 V
13	10	18	DVSUP		X	Digital power supply +5 V
14	–	–	DVSS		X	Digital ground
15	–	–	DVSS		X	Digital ground
16	11	19	DVSS		X	Digital ground
–	12	20	I2S_DA_IN2/3	IN	LV	I ² S2/3-data input
17	–	–	I2S_DA_IN2	IN	LV	PQFP80: pin 22 separate I2S_DA_IN3
18	13	21	NC		LV	Not connected
19	14	22	I2S_CL3	IN	LV	I ² S3 clock
20	15	23	I2S_WS3	IN	LV	I ² S3 word strobe
21	16	24	RESETQ	IN	X	Power-on-reset
22	–	–	I2S_DA_IN3	IN	LV	I ² S3-data input
23	–	–	NC		LV	Not connected
24	17	25	DACA_R	OUT	LV	Aux out, right
25	18	26	DACA_L	OUT	LV	Aux out, left
26	19	27	VREF2		X	Reference ground 2

PQFP 80-pin	Pin No.		Pin Name	Type	Connection (if not used)	Short Description
	PLQFP 64-pin	PSDIP 64-pin				
27	20	28	DACM_R	OUT	LV	Main out, right
28	21	29	DACM_L	OUT	LV	Main out, left
29	22	30	NC		LV	Not connected
30	23	31	NC		LV	Not connected
31	24	32	NC		LV	Not connected
32	–	–	NC		LV	Not connected
33	25	33	SC2_OUT_R	OUT	LV	SCART output 2, right
34	26	34	SC2_OUT_L	OUT	LV	SCART output 2, left
35	27	35	VREF1		X	Reference ground 1
36	28	36	SC1_OUT_R	OUT	LV	SCART output 1, right
37	29	37	SC1_OUT_L	OUT	LV	SCART output 1, left
38	30	38	CAPL_A		X	Volume capacitor Aux
39	31	39	AHVSUP		X	Analog power supply 8.0 V
40	32	40	CAPL_M		X	Volume capacitor Main
41	–	–	NC		LV	Not connected
42	–	–	NC		LV	Not connected
43	–	–	AHVSS		X	Analog ground
44	33	41	AHVSS		X	Analog ground
45	34	42	AGNDC		X	Analog reference voltage
46	–	–	NC		LV	Not connected
47	35	43	SC4_IN_L	IN	LV	SCART 4 input, left
48	36	44	SC4_IN_R	IN	LV	SCART 4 input, right
49	37	45	ASG		AHVSS	Analog Shield Ground
50	38	46	SC3_IN_L	IN	LV	SCART 3 input, left
51	39	47	SC3_IN_R	IN	LV	SCART 3 input, right
52	40	48	ASG		AHVSS	Analog Shield Ground
53	41	49	SC2_IN_L	IN	LV	SCART 2 input, left
54	42	50	SC2_IN_R	IN	LV	SCART 2 input, right
55	43	51	ASG		AHVSS	Analog Shield Ground
56	44	52	SC1_IN_L	IN	LV	SCART 1 input, left
57	45	53	SC1_IN_R	IN	LV	SCART 1 input, right

PQFP 80-pin	Pin No.		Pin Name	Type	Connection (if not used)	Short Description
	PLQFP 64-pin	PSDIP 64-pin				
58	46	54	VREFTOP		X	Reference voltage IF A/D converter
59	–	–	NC		LV	Not connected
60	47	55	MONO_IN	IN	LV	Mono input
61	–	–	AVSS		X	Analog ground
62	48	56	AVSS		X	Analog ground
63	–	–	NC		LV	Not connected
64	–	–	NC		LV	Not connected
65	–	–	AVSUP		X	Analog power supply +5 V
66	49	57	AVSUP		X	Analog power supply +5 V
67	50	58	ANA_IN1+	IN	LV	IF input 1
68	51	59	ANA_IN–	IN	AVSS via 56 pF / LV	IF common (Can be left vacant, only if IF input 1 is also not in use)
69	52	60	ANA_IN2+	IN	AVSS via 56 pF / LV	IF input 2 (Can be left vacant, only if IF input 1 is also not in use)
70	53	61	TESTEN	IN	AVSS	Test pin
71	54	62	XTAL_IN	IN	X	Crystal oscillator
72	55	63	XTAL_OUT	OUT	X / LV	Crystal oscillator (See also Section 4.3. “Pin Descriptions” on page 42)
73	56	64	TP		LV	Test pin
74	57	1	AUD_CL_OUT	OUT	LV	Audio clock output (18.432 MHz)
75	58	2	NC		LV	Not connected
76	59	3	NC		LV	Not connected
77	60	4	D_CTR_I/O_1	IN/OUT	LV	D_CTR_I/O_1
78	61	5	D_CTR_I/O_0	IN/OUT	LV	D_CTR_I/O_0
79	62	6	ADR_SEL	IN	X	I ² C Bus address select
80	63	7	STANDBYQ	IN	X	Stand-by (low-active)

4.3. Pin Descriptions

Pin numbers refer to the 80-pin PQFP package.

Pin 1, **NC** – Pin not connected.

Pin 2, **I2C_CL** – I²C Clock Input/Output (Fig. 4–8)
Via this pin, the I²C-bus clock signal has to be supplied. The signal can be pulled down by the MSP in case of wait conditions.

Pin 3, **I2C_DA** – I²C Data Input/Output (Fig. 4–8)
Via this pin, the I²C-bus data is written to or read from the MSP.

Pin 4, **I2S_CL** – I²S Clock Input/Output (Fig. 4–11)
Clock line for the synchronous I²S bus. In master mode, this line is driven by the MSP; in slave mode, an external I²S clock has to be supplied.

Pin 5, **I2S_WS** – I²S Word Strobe Input/Output (Fig. 4–11)
Word strobe line for the synchronous I²S bus. In master mode, this line is driven by the MSP; in slave mode, an external I²S word strobe has to be supplied.

Pin 6, **I2S_DA_OUT1** – I²S Data Output (Fig. 4–7)
Output of digital serial sound data of the MSP on the synchronous I²S bus.

Pin 7, **I2S_DA_IN1** – I²S Data Input 1 (Fig. 4–9)
First input of digital serial sound data to the MSP via the synchronous I²S bus.

Pin 8, **ADR_DA** – ADR Bus Data Output (Fig. 4–7)
Output of digital serial data to the DRP 3510A via the ADR bus.

Pin 9, **ADR_WS** – ADR Bus Word Strobe Output (Fig. 4–7)
Word strobe output for the ADR bus.

Pin 10, **ADR_CL** – ADR Bus Clock Output (Fig. 4–7)
Clock line for the ADR bus.

Pins 11, 12, 13, **DVSUP*** – Digital Supply Voltage
Power supply for the digital circuitry of the MSP. Must be connected to a +5 V power supply.

Pins 14, 15, 16, **DVSS*** – Digital Ground
Ground connection for the digital circuitry of the MSP.

Pin 17, **I2S_DA_IN2** – I²S Data Input 2 (Fig. 4–9)
Second input of digital serial sound data to the MSP via the synchronous I²S bus. In all packages except PQFP80, this pin is also connected to the asynchronous I²S interface 3.

Pins 18, **NC** – Pin not connected.

Pins 19, **I2S_CL3** – I²S Clock Input (Fig. 4–9)
Clock line for the asynchronous I²S bus. Since only a slave mode is available an external I²S clock has to be supplied.

Pins 20, **I2S_WS3** – I²S Word Strobe Input (Fig. 4–9)
Word strobe line for the asynchronous I²S bus. Since only a slave mode is available an external I²S word strobe has to be supplied.

Pin 21, **RESETQ** – Reset Input (Fig. 4–9)
In the steady state, high level is required. A low level resets the MSP 44x8G.

Pin 22, **I2S_DA_IN3** – I²S Data Input 3 (Fig. 4–9)
Input of digital serial sound data to the MSP via the asynchronous I²S bus. In all packages except PQFP80, this pin is also connected to synchronous I²S interface 2.

Pins 23, **NC** – Pin not connected.

Pins 24, 25, **DACA_R/L** – Aux Outputs (Fig. 4–17)
Output of the Aux signal. A 1 nF capacitor to AHVSS must be connected to these pins. The DC offset on these pins depends on the selected Aux volume.

Pin 26, **VREF2** – Reference Ground 2
Reference analog ground. This pin must be connected separately to the ground (AHVSS). VREF2 serves as a clean ground and should be used as the reference for analog connections to the Main and Aux outputs.

Pins 27, 28, **DACM_R/L** – Main Outputs (Fig. 4–17)
Output of the Main signal. A 1 nF capacitor to AHVSS must be connected to these pins. The DC offset on these pins depends on the selected Main volume.

Pin 29, 30, 31, 32 **NC** – Pin not connected.

Pins 33, 34, **SC2_OUT_R/L** – SCART2 Outputs (Fig. 4–19)
Output of the SCART2 signal. Connections to these pins must use a 100-Ω series resistor and are intended to be AC-coupled.

Pin 35, **VREF1** – Reference Ground 1
Reference analog ground. This pin must be connected separately to the ground (AHVSS). VREF1 serves as a clean ground and should be used as the reference for analog connections to the SCART outputs.

Pins 36, 37, **SC1_OUT_R/L** – SCART1 Outputs (Fig. 4–19)
Output of the SCART1 signal. Connections to these pins must use a 100-Ω series resistor and are intended to be AC-coupled.

Pin 38, **CAPLA** – Volume Capacitor Aux (Fig. 4–14)
A 10- μ F capacitor to AHVSUP must be connected to this pin. It serves as a smoothing filter for Aux volume changes in order to suppress audible plops. The value of the capacitor can be lowered to 1- μ F if faster response is required. The area encircled by the trace lines should be minimized; keep traces as short as possible. This input is sensitive for magnetic induction.

Pin 39, **AHVSUP*** – Analog Power Supply High Voltage

Power is supplied via this pin for the analog circuitry of the MSP (except IF input). This pin must be connected to the +8 V supply. (+5 V-operation is possible with restrictions in performance)

Pin 40, **CAPLM** – Volume Capacitor Main (Fig. 4–14)
A 10- μ F capacitor to AHVSUP must be connected to this pin. It serves as a smoothing filter for Main volume changes in order to suppress audible plops. The value of the capacitor can be lowered to 1 μ F if faster response is required. The area encircled by the trace lines should be minimized; keep traces as short as possible. This input is sensitive for magnetic induction.

Pins 41, 42, **NC** – Pins not connected.

Pins 43, 44, **AHVSS*** – Analog Power Supply High Voltage

Ground connection for the analog circuitry of the MSP (except IF input).

Pin 45, **AGNDC** – Internal Analog Reference Voltage
This pin serves as the internal ground connection for the analog circuitry (except IF input). It must be connected to the VREF pins with a 3.3- μ F and a 100-nF capacitor in parallel. This pins shows a DC level of typically 3.73 V.

Pin 46, **NC** – Pin not connected.

Pins 47, 48, **SC4_IN_L/R** – SCART4 Inputs (Fig. 4–16)

The analog input signal for SCART4 is fed to this pin. Analog input connection must be AC-coupled.

Pins 49, 52, and 55, **ASG*** – Analog Shield Ground
Analog ground (AHVSS) should be connected to this pin to reduce cross-coupling between SCART inputs.

Pins 50, 51, **SC3_IN_L/R** – SCART3 Inputs (Fig. 4–16)

The analog input signal for SCART3 is fed to this pin. Analog input connection must be AC-coupled.

Pins 53, 54 **SC2_IN_L/R** – SCART2 Inputs (Fig. 4–16)
The analog input signal for SCART2 is fed to this pin. Analog input connection must be AC-coupled.

Pins 56, 57 **SC1_IN_L/R** – SCART1 Inputs (Fig. 4–16)
The analog input signal for SCART1 is fed to this pin. Analog input connection must be AC-coupled.

Pin 58, **VREFTOP** – Reference Voltage IF A/D Converter (Fig. 4–13)

Via this pin, the reference voltage for the IF A/D converter is decoupled. It must be connected to AVSS pins with a 10- μ F and a 100-nF capacitor in parallel. Traces must be kept short.

Pin 59, **NC** – Pin not connected.

Pin 60 **MONO_IN** – Mono Input (Fig. 4–16)

The analog mono input signal is fed to this pin. Analog input connection must be AC-coupled.

Pins 61, 62, **AVSS*** – Analog Power Supply Voltage
Ground connection for the analog IF input circuitry of the MSP.

Pins 63, 64, **NC** – Pins not connected.

Pins 65, 66, **AVSUP*** – Analog Power Supply Voltage
Power is supplied via this pin for the analog IF input circuitry of the MSP. This pin must be connected to the +5 V supply.

Pin 67, **ANA_IN1+** – IF Input 1 (Fig. 4–13)

The analog sound IF signal is supplied to this pin. Inputs must be AC-coupled. This pin is designed as symmetrical input: ANA_IN1+ is internally connected to one input of a symmetrical op amp, ANA_IN- to the other.

Pin 68, **ANA_IN-** – IF Common (Fig. 4–13)

This pins serves as a common reference for ANA_IN1/2+ inputs and must be AC-coupled.

Pin 69, **ANA_IN2+** – IF Input 2 (Fig. 4–13)

The analog sound if signal is supplied to this pin. Inputs must be AC-coupled. This pin is designed as symmetrical input: ANA_IN2+ is internally connected to one input of a symmetrical op amp, ANA_IN- to the other.

Pin 70, **TESTEN** – Test Enable Pin (Fig. 4–9)

This pin enables factory test modes. For normal operation, it must be connected to ground.

Pins 71, 72 **XTAL_IN, XTAL_OUT** – Crystal Input and Output Pins (Fig. 4–12)

These pins are connected to an 18.432 MHz crystal oscillator which is digitally tuned by integrated capacitances. An external clock can be fed into XTAL_IN (leave XTAL_OUT vacant in this case). The audio clock output signal AUD_CL_OUT is derived from the oscillator. External capacitors at each crystal pin to ground (AVSS) are required. It should be verified by layout, that no supply current for the digital circuitry is flowing through the ground connection point.

Pin 73, **TP** – This pin enables factory test modes. For normal operation, it must be left vacant.

Pin 74, **AUD_CL_OUT** – Audio Clock Output (Fig. 4–12)

This is the 18.432 MHz main clock output.

Pins 75, 76, **NC** – Pins not connected.

Pins 77, 78, **D_CTR_I/O_1/0** – Digital Control Input/Output Pins (Fig. 4–11)

These pins serve as general purpose input/output pins. Pin D_CTR_I/O_1 can be used as an interrupt request pin to the controller.

Pin 79, **ADR_SEL** – I²C Bus Address Select (Fig. 4–10)

By means of this pin, one of three device addresses for the MSP can be selected. The pin can be connected to ground (I²C device addresses 80/81_{hex}), to +5 V supply (84/85_{hex}), or left open (88/89_{hex}).

Pin 80, **STANDBYQ** – Stand-by

In normal operation, this pin must be High. If the MSP is switched off by first pulling STANDBYQ low and then (after >1 μs delay) switching off the 5 V, but keeping the 8-V power supply ('**Stand-by**'-mode), the SCART switches maintain their position and function.

Pin -, **I2S_DA_IN2/3** – I²S data input (see Fig. 4–9).

This pin is connected to I2S_DA_IN2 and I2S_DA_IN3. Not available for PQFP80-pin package.

*** Application Note:**

All ground pins should be connected to one low-resistive ground plane.

All supply pins should be connected separately with short and low-resistive lines to the power supply.

Decoupling capacitors from DVSUP to DVSS, AVSUP to AVSS, and AHVSUP to AHVSS are recommended as closely as possible to these pins. Decoupling of DVSUP and DVSS is most important. We recommend using more than one capacitor. By choosing different values, the frequency range of active decoupling can be extended. In our application boards we use: 220 pF, 470 pF, 1.5 nF, and 10 μF. The capacitor with the lowest value should be placed nearest to the pins.

The ASG pins should be connected as closely as possible to the MSP ground. They are intended for leading with the SCART signals as shield lines and should not be connected to ground at the SCART-connector.

4.4. Pin Configurations

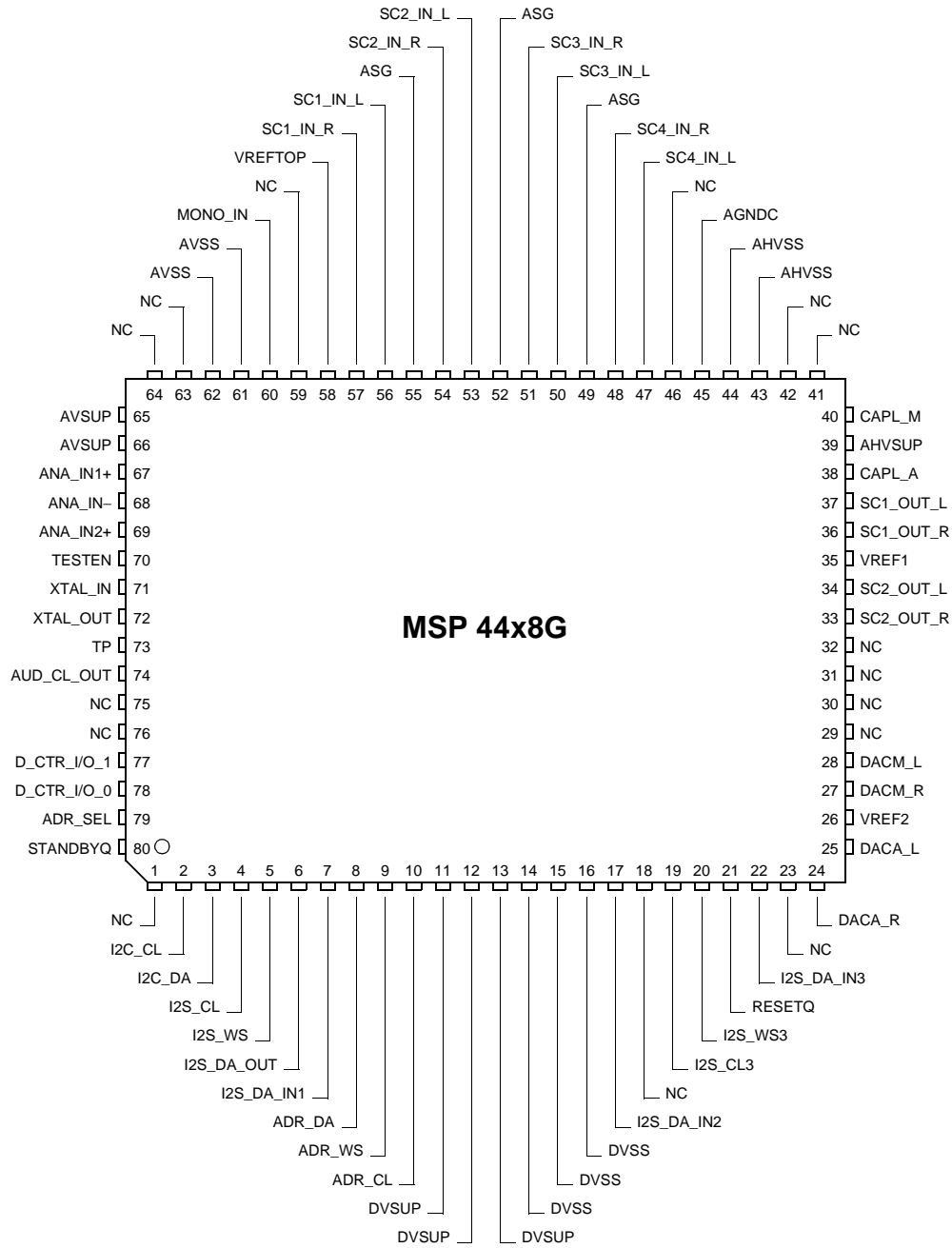


Fig. 4-4: 80-pin PQFP package

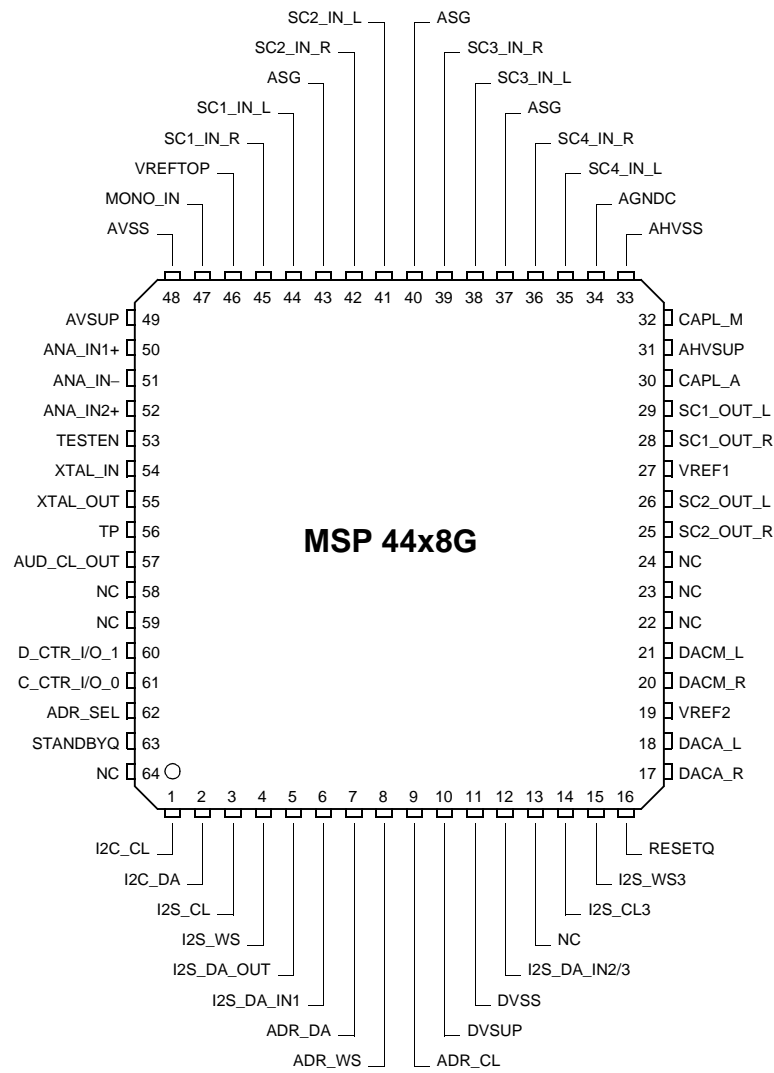


Fig. 4-5: 64-pin PLQFP package

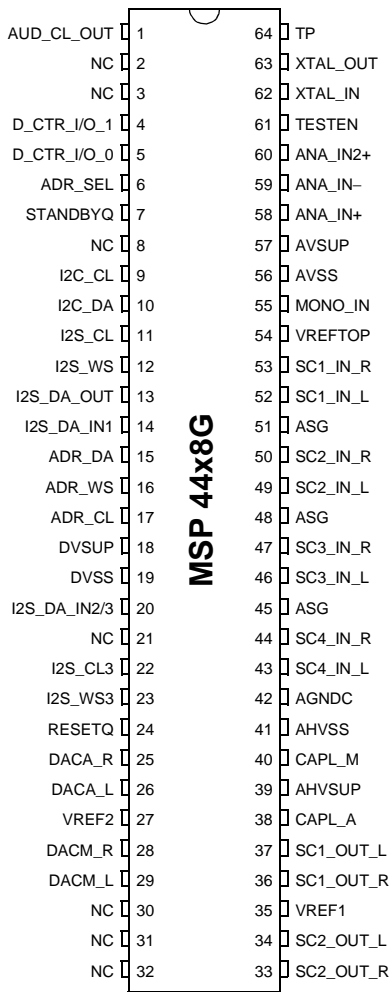


Fig. 4–6: 64-pin PSDIP package

4.5. Pin Circuits

Pin numbers refer to the PQFP80 package.

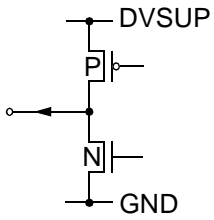


Fig. 4-7: Output Pins 6, 8, 9, and 10 (I2S_DA_OUT, ADR_DA, ADR_WS, ADR_CL)

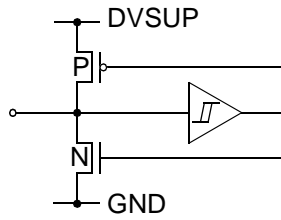


Fig. 4-11: Input/Output Pins 4, 5, 77, and 78 (I2S_CL, I2S_WS, D_CTR_I/O_1, D_CTR_I/O_0)

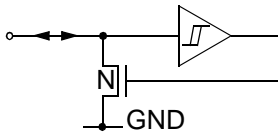


Fig. 4-8: Input/Output Pins 2 and 3 (I2C_CL, I2C_DA)

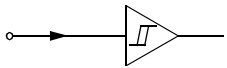


Fig. 4-9: Input Pins 7, 17, 22, 19, 20, 21, 70, and 80 (I2S_DA_IN1..3, I2S_CL3, I2S_WS3, RESETQ, TESTEN, STANDBYQ)

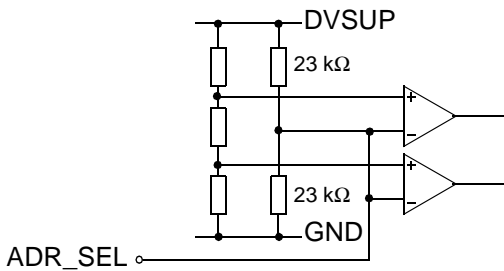


Fig. 4-10: Input Pin 79 (ADR_SEL)

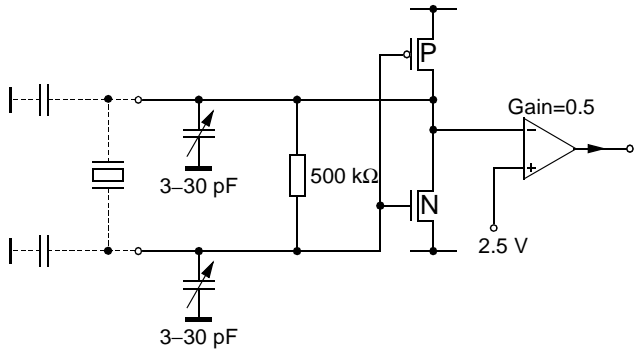


Fig. 4-12: Output/Input Pins 71, 72, and 74 (XTAL_IN, XTAL_OUT, AUD_CL_OUT)

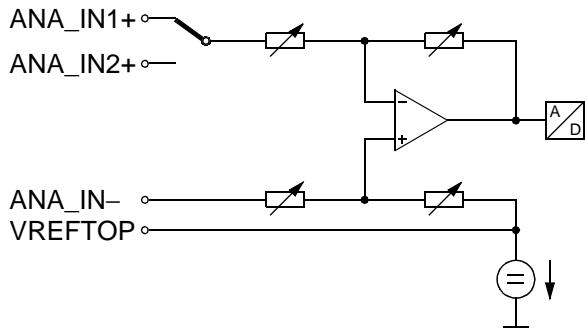


Fig. 4-13: Input Pins 58, 67, 68, and 69 (VREFTOP, ANA_IN1+, ANA_IN-, ANA_IN2+)

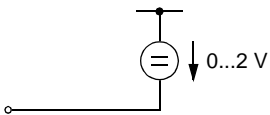


Fig. 4-14: Capacitor Pins 38 and 40 (CAPL_A, CAPL_M)

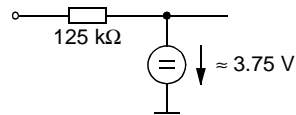


Fig. 4-18: Pin 45 (AGNDC)

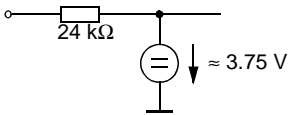


Fig. 4-15: Input Pin 60 (MONO_IN)

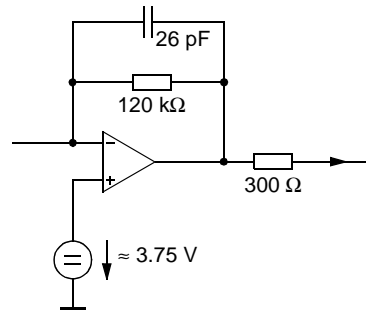


Fig. 4-19: Output Pins 33, 34, 36, and 37 (SC_2_OUT_R/L, SC_1_OUT_R/L)

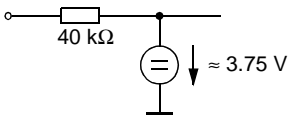


Fig. 4-16: Input Pins 47, 48, 50, 51, 53, 54, 56, and 57 (SC4-1_IN_L/R)

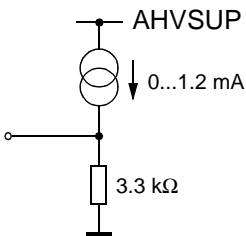


Fig. 4-17: Output Pins 24, 25, 27, and 28 (DACA_R/L, DACM_R/L)

4.6. Electrical Characteristics

4.6.1. Absolute Maximum Ratings

Symbol	Parameter	Pin Name	Min.	Max.	Unit
T_A	Ambient Operating Temperature	–	0	70 ¹⁾	°C
T_S	Storage Temperature	–	–40	125	°C
V_{SUP1}	First Supply Voltage	AHVSUP	–0.3	9.0	V
V_{SUP2}	Second Supply Voltage	DVSUP	–0.3	6.0	V
V_{SUP3}	Third Supply Voltage	AVSUP	–0.3	6.0	V
dV_{SUP23}	Voltage between AVSUP and DVSUP	AVSUP, DVSUP	–0.5	0.5	V
P_{TOT}	Package Power Dissipation PSDIP64 PLQFP64 PQFP80	AHVSUP, DVSUP, AVSUP		1300 960 1000	mW mW mW
V_{Idig}	Input Voltage, all Digital Inputs		–0.3	$V_{SUP2}+0.3$	V
I_{Idig}	Input Current, all Digital Pins		–20	+20	mA ²⁾
V_{Iana}	Input Voltage, all Analog Inputs	SCn_IN_s, ³⁾ MONO_IN	–0.3	$V_{SUP1}+0.3$	V
I_{Iana}	Input Current, all Analog Inputs	SCn_IN_s, ³⁾ MONO_IN	–5	+5	mA ²⁾
I_{Oana}	Output Current, all SCART Outputs	SCn_OUT_s ³⁾	4), 5)	4), 5)	
I_{Oana}	Output Current, all Analog Outputs except SCART Outputs	DACp_s ³⁾	4)	4)	
I_{Cana}	Output Current, other pins connected to capacitors	CAPL_p, ³⁾ AGNDC	4)	4)	
<p>1) PLQFP64: 65 °C 2) positive value means current flowing into the circuit 3) “n” means “1”, “2”, “3”, or “4”, “s” means “L” or “R”, “p” means “M” or “A” 4) The Analog Outputs are short-circuit proof with respect to First Supply Voltage and Ground. 5) Total chip power dissipation must not exceed absolute maximum rating.</p>					

Stresses beyond those listed in the “Absolute Maximum Ratings” may cause permanent damage to the device. This is a stress rating only. Functional operation of the device at these or any other conditions beyond those indicated in the “Recommended Operating Conditions/Characteristics” of this specification is not implied. Exposure to absolute maximum ratings conditions for extended periods may affect device reliability.

4.6.2. Recommended Operating Conditions ($T_A = 0$ to 70 °C)**4.6.2.1. General Recommended Operating Conditions**

Symbol	Parameter	Pin Name	Min.	Typ.	Max.	Unit
V_{SUP1}	First Supply Voltage (8-V Operation)	AHVSUP	7.6	8.0	8.7	V
	First Supply Voltage (5-V Operation)		4.75	5.0	5.25	V
V_{SUP2}	Second Supply Voltage (5-V Operation)	DVSUP	4.75	5.0	5.25	V
	Second Supply Voltage (3.3-V Operation)		3.15	3.3	3.45	V
V_{SUP3}	Third Supply Voltage	AVSUP	4.75	5.0	5.25	V
t_{STBYQ1}	STANDBYQ Setup Time before Turn-off of Second Supply Voltage	STANDBYQ, DVSUP	1			μ s

4.6.2.2. Analog Input and Output Recommendations

Symbol	Parameter	Pin Name	Min.	Typ.	Max.	Unit
C_{AGNDC}	AGNDC-Filter-Capacitor	AGNDC	-20%	3.3		μ F
	Ceramic Capacitor in Parallel		-20%	100		nF
C_{inSC}	DC-Decoupling Capacitor in front of SCART Inputs	$SCn_IN_s^1)$	-20%	330		nF
V_{inSC}	SCART Input Level				2.0	V_{RMS}
V_{inMONO}	Input Level, Mono Input	MONO_IN			2.0	V_{RMS}
R_{LSC}	SCART Load Resistance	$SCn_OUT_s^1)$	10			k Ω
C_{LSC}	SCART Load Capacitance				6.0	nF
C_{VMA}	Main/Aux Volume Capacitor	CAPL_M, CAPL_A		10		μ F
C_{FMA}	Main/Aux Filter Capacitor	DACM_s, DACA_s ¹⁾	-10%	1	+10%	nF

1) "n" means "1", "2", or "3", "s" means "L" or "R", "p" means "M" or "A"

4.6.2.3. Recommendations for Analog Sound IF Input Signal

Symbol	Parameter	Pin Name	Min.	Typ.	Max.	Unit
C _{VREFTOP}	VREFTOP-Filter-Capacitor	VREFTOP	-20 %	10		μF
	Ceramic Capacitor in Parallel		-20 %	100		nF
F _{IF_FMTV}	Analog Input Frequency Range for TV Applications	ANA_IN1+, ANA_IN2+, ANA_IN-	0		9	MHz
F _{IF_FMRADIO}	Analog Input Frequency for FM-Radio Applications			10.7		MHz
V _{IF_FM}	Analog Input Range FM/NICAM		0.1	0.8	3	V _{pp}
V _{IF_AM}	Analog Input Range AM/NICAM		0.1	0.45	0.8	V _{pp}
R _{FMNI}	Ratio: NICAM Carrier/FM Carrier (unmodulated carriers) BG: I:		-20	-7	0	dB
			-23	-10	0	
R _{AMNI}	Ratio: NICAM Carrier/AM Carrier (unmodulated carriers)		-25	-11	0	dB
R _{FM}	Ratio: FM-Main/FM-Sub Satellite			7		dB
R _{FM1/FM2}	Ratio: FM1/FM2 German FM-System			7		dB
R _{FC}	Ratio: Main FM Carrier/ Color Carrier		15	-	-	dB
R _{FV}	Ratio: Main FM Carrier/ Luma Components		15	-	-	dB
PR _{IF}	Passband Ripple		-	-	±2	dB
SUP _{HF}	Suppression of Spectrum above 9.0 MHz (not for FM Radio)		15			dB
FM _{MAX}	Maximum FM-Deviation (approx.) normal mode HDEV2: high deviation mode HDEV3: very high deviation mode				±180	kHz
				±360		
				±540		

4.6.2.4. Crystal Recommendations

Symbol	Parameter	Pin Name	Min.	Typ.	Max.	Unit
General Crystal Recommendations						
f_P	Crystal Parallel Resonance Frequency at 12 pF Load Capacitance			18.432		MHz
R_R	Crystal Series Resistance			8	25	Ω
C_0	Crystal Shunt (Parallel) Capacitance			6.2	7.0	pF
C_L	External Load Capacitance ¹⁾	XTAL_IN, XTAL_OUT	PSDIP approx. 1.5 P(L)QFP approx. 3.3			pF pF
Crystal Recommendations for Master-Slave Applications (MSP-clock must perform synchronization to I ² S clock)						
f_{TOL}	Accuracy of Adjustment		-20		+20	ppm
D_{TEM}	Frequency Variation versus Temperature		-20		+20	ppm
C_1	Motional (Dynamic) Capacitance		19	24		fF
f_{CL}	Required Open Loop Clock Frequency ($T_{amb} = 25\text{ }^\circ\text{C}$)	AUD_CL_OUT	18.431		18.433	MHz
Crystal Recommendations for FM / NICAM Applications (No MSP-clock synchronization to I ² S clock possible)						
f_{TOL}	Accuracy of Adjustment		-30		+30	ppm
D_{TEM}	Frequency Variation versus Temperature		-30		+30	ppm
C_1	Motional (Dynamic) Capacitance		15			fF
f_{CL}	Required Open Loop Clock Frequency ($T_{amb} = 25\text{ }^\circ\text{C}$)	AUD_CL_OUT	18.4305		18.4335	MHz
Crystal Recommendations for all analog FM/AM Applications (No MSP-clock synchronization to I ² S clock possible)						
f_{TOL}	Accuracy of Adjustment		-100		+100	ppm
D_{TEM}	Frequency Variation versus Temperature		-50		+50	ppm
f_{CL}	Required Open Loop Clock Frequency ($T_{amb} = 25\text{ }^\circ\text{C}$)	AUD_CL_OUT	18.429		18.435	MHz
Amplitude Recommendation for Operation with External Clock Input (C_{load} after reset typ. 22 pF)						
V_{XCA}	External Clock Amplitude	XTAL_IN	0.7			V_{pp}
<p>¹⁾ External capacitors at each crystal pin to ground are required. They are necessary to tune the open-loop frequency of the internal PLL and to stabilize the frequency in closed-loop operation. Due to different layouts, <u>the accurate capacitor size should be determined with the customer PCB</u>. The suggested values (1.5...3.3 pF) are figures based on experience and should serve as "start value".</p> <p>To define the capacitor size, reset the MSP without transmitting any further I2C telegrams. Measure the frequency at AUD_CL_OUT-pin. Change the capacitor size until the free running frequency matches 18.432 MHz as closely as possible. The higher the capacity, the lower the resulting clock frequency.</p>						

4.6.3. Characteristics

at $T_A = 0$ to 70 °C, $f_{\text{CLOCK}} = 18.432$ MHz, $V_{\text{SUP1}} = 7.6$ to 8.7 V, $V_{\text{SUP2}} = 4.75$ to 5.25 V for min./max. values

at $T_A = 60$ °C, $f_{\text{CLOCK}} = 18.432$ MHz, $V_{\text{SUP1}} = 8$ V, $V_{\text{SUP2}} = 5$ V for typical values,

T_j = Junction Temperature

Main (M) = Main Channel, Aux (A) = Aux Channel

4.6.3.1. General Characteristics

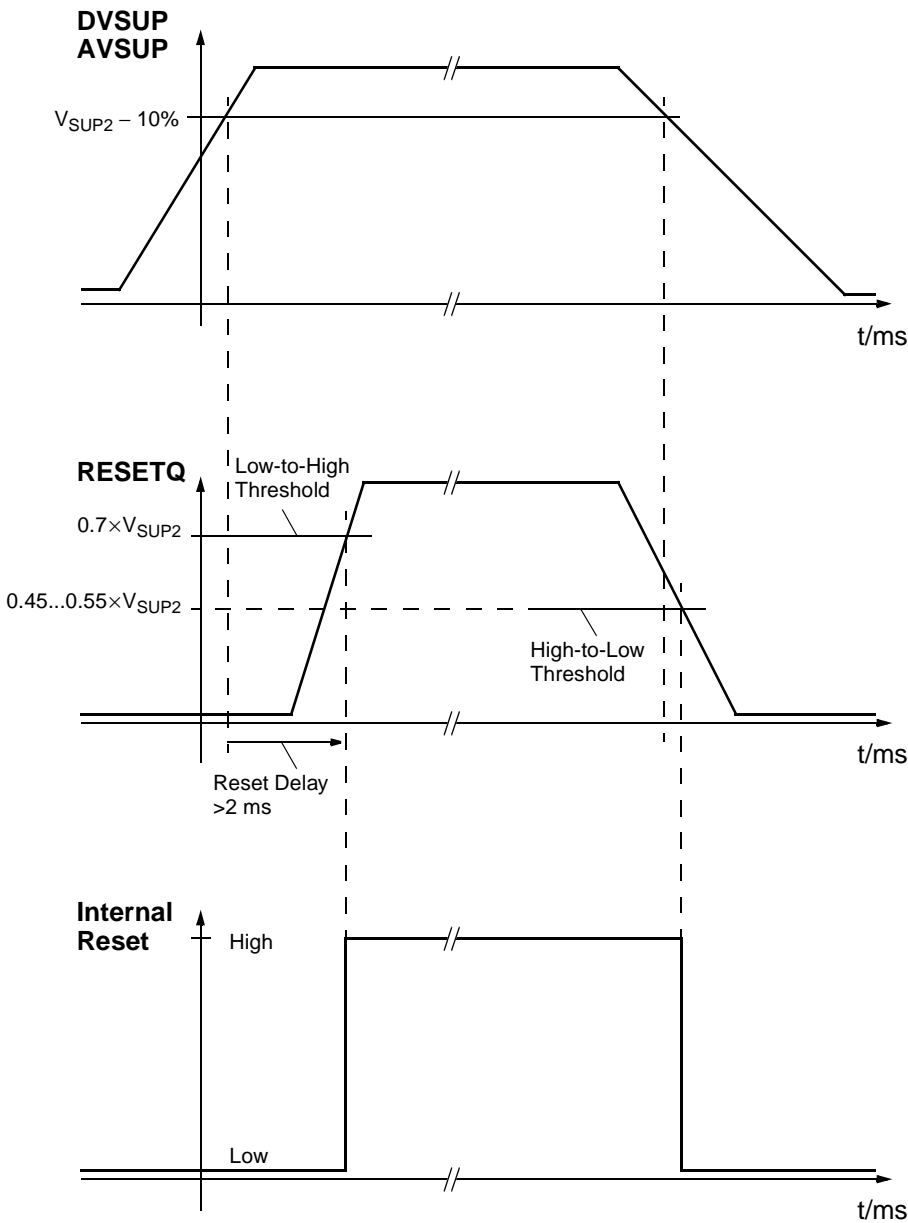
Symbol	Parameter	Pin Name	Min.	Typ.	Max.	Unit	Test Conditions
Supply							
I_{SUP1A}	First Supply Current (active) (8-V Operation) Analog Volume for Main and Aux at 0 dB Analog Volume for Main and Aux at -30 dB	AHVSUP		18 12	25 17	mA mA	
	First Supply Current (active) (5-V Operation) Analog Volume for Main and Aux at 0 dB Analog Volume for Main and Aux at -30 dB			12 8	17 11	mA mA	
I_{SUP2A}	Second Supply Current (active) (5-V Operation)	DVSUP		70	85	mA	
	Second Supply Current (active) (3.3-V Operation)			60	75	mA	
I_{SUP3A}	Third Supply Current (active)	AVSUP		35	45	mA	
I_{SUP1S}	First Supply Current (8-V Operation) (standby mode) at $T_j = 27$ °C	AHVSUP		5.6	7.7	mA	STANDBYQ = low
	First Supply Current (5-V Operation) (standby mode) at $T_j = 27$ °C			3.7	5.1	mA	STANDBYQ = low
Clock							
f_{CLOCK}	Clock Input Frequency	XTAL_IN		18.432		MHz	
D_{CLOCK}	Clock High to Low Ratio			45	55	%	
t_{JITTER}	Clock Jitter (Verification not provided in Production Test)				50	ps	
V_{xtalDC}	DC-Voltage Oscillator			2.5		V	
t_{Startup}	Oscillator Startup Time at VDD Slew-rate of 1 V/ μ s	XTAL_IN, XTAL_OUT		0.4	2	ms	
V_{ACLKAC}	Audio Clock Output AC Voltage	AUD_CL_OUT	1.2	1.8		V_{pp}	load = 40 pF
V_{ACLKDC}	Audio Clock Output DC Voltage			0.4	0.6	V_{SUP3}	$I_{\text{max}} = 0.2$ mA
$r_{\text{outHF_ACL}}$	HF Output Resistance				140	Ω	

4.6.3.2. Digital Inputs, Digital Outputs

Symbol	Parameter	Pin Name	Min.	Typ.	Max.	Unit	Test Conditions
Digital Inputs Levels							
V _{DIGIL}	Digital Input Low Voltage	STANDBYQ D_CTR_I/O_0/1			0.2	V _{SUP2}	
V _{DIGIH}	Digital Input High Voltage		0.5			V _{SUP2}	
Z _{DIGI}	Input Impedance				5	pF	
I _{DLEAK}	Digital Input Leakage Current		-1		1	μA	0 V < U _{INPUT} < DV _{SUP} D_CTR_I/O_0/1: tri-state
V _{DIGIL}	ADR_SEL Input Low Voltage	ADR_SEL			0.2	V _{SUP2}	
V _{DIGIH}	ADR_SEL Input High Voltage		0.8			V _{SUP2}	
I _{ADRSEL}	Input Current		-500	-220		μA	U _{ADR_SEL} = DV _{SS}
				220	500	μA	U _{ADR_SEL} = DV _{SUP}
Digital Output Levels							
V _{DCTROL}	Digital Output Low Voltage	D_CTR_I/O_0 D_CTR_I/O_1			0.4	V	ID _{DCTR} = 1 mA
V _{DCTROH}	Digital Output High Voltage		V _{SUP2} -0.3			V	ID _{DCTR} = -1 mA

4.6.3.3. Reset Input and Power-Up

Symbol	Parameter	Pin Name	Min.	Typ.	Max.	Unit	Test Conditions
RESETQ Input Levels							
V_{RHL}	Reset High-Low Transition Voltage	RESETQ	0.45		0.55	V_{SUP2}	
V_{RLH}	Reset Low-High Transition Voltage		0.7		0.8	V_{SUP2}	
Z_{RES}	Input Impedance				5	pF	
I_{RES}	Input Pin Leakage Current		-1		1	μA	$0 V < U_{INPUT} < DVSUP$



Note: The reset should not reach high level before the oscillator has started. This requires a reset delay of >2 ms

$0.7 \times V_{SUP2}$ means 3.5 Volt with $V_{SUP2} = 5.0 V$

Fig. 4-20: Power-up sequence

4.6.3.4. I²C-Bus Characteristics

Symbol	Parameter	Pin Name	Min.	Typ.	Max.	Unit	Test Conditions
V _{I2CIL}	I ² C-BUS Input Low Voltage	I2C_CL, I2C_DA			0.3	V _{SUP2}	
V _{I2CIH}	I ² C-BUS Input High Voltage		0.6			V _{SUP2}	
t _{I2C1}	I ² C START Condition Setup Time		120			ns	
t _{I2C2}	I ² C STOP Condition Setup Time		120			ns	
t _{I2C5}	I ² C-Data Setup Time before Rising Edge of Clock		55			ns	
t _{I2C6}	I ² C-Data Hold Time after Falling Edge of Clock		55			ns	
t _{I2C3}	I ² C-Clock Low Pulse Time	I2C_CL	500			ns	
t _{I2C4}	I ² C-Clock High Pulse Time		500			ns	
f _{I2C}	I ² C-BUS Frequency				1.0	MHz	
V _{I2COL}	I ² C-Data Output Low Voltage	I2C_CL, I2C_DA			0.4	V	I _{I2COL} = 3 mA
I _{I2COH}	I ² C-Data Output High Leakage Current				1.0	μA	V _{I2COH} = 5 V
t _{I2COL1}	I ² C-Data Output Hold Time after Falling Edge of Clock		15			ns	
t _{I2COL2}	I ² C-Data Output Setup Time before Rising Edge of Clock		100			ns	f _{I2C} = 1 MHz

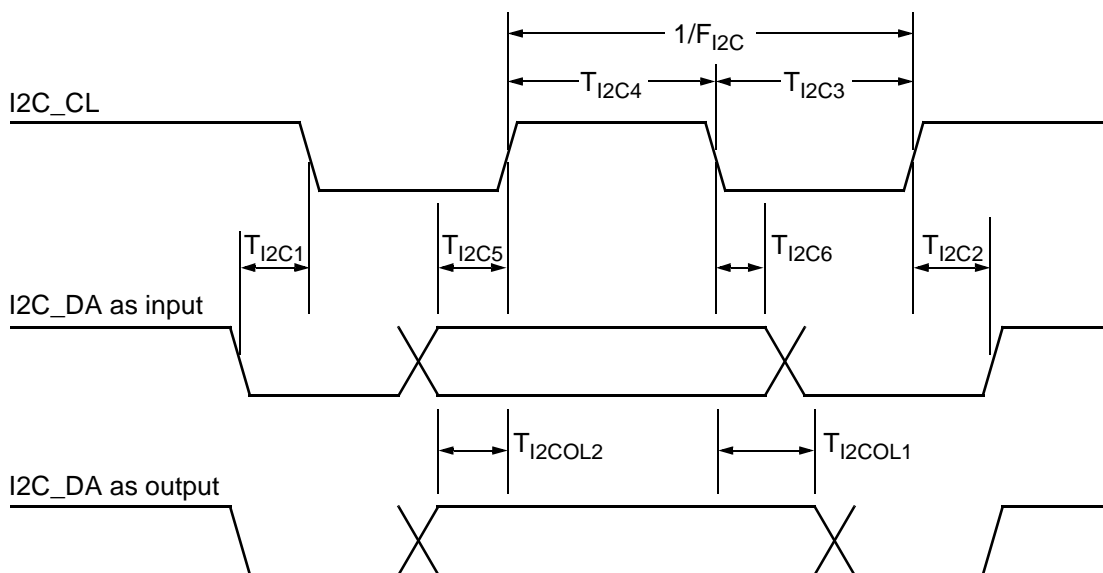
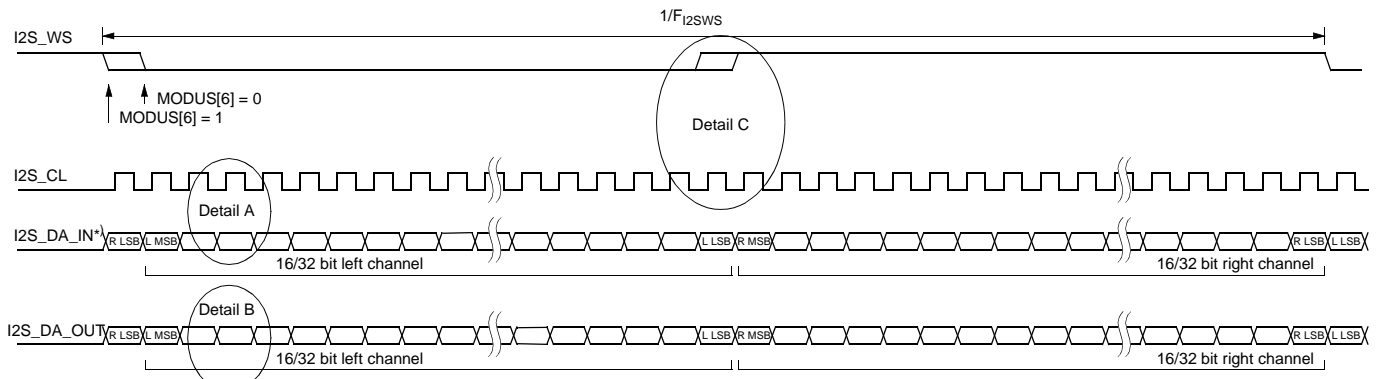


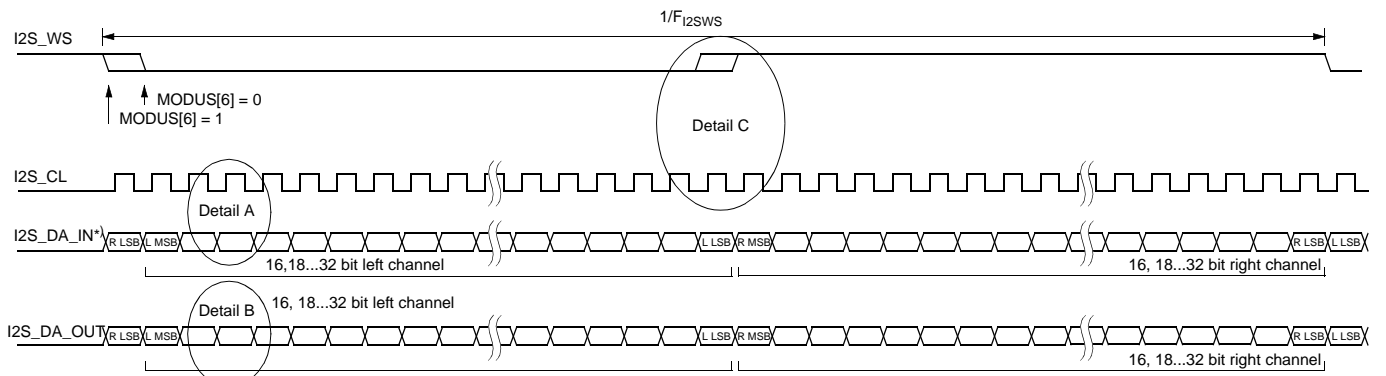
Fig. 4–21: I²C bus timing diagram

4.6.3.5. I²S-Bus Characteristics

Symbol	Parameter	Pin Name	Min.	Typ.	Max.	Unit	Test Conditions
V _{I2SIL}	Input Low Voltage	I2S_CL I2S_WS I2S_CL3 I2S_WS3 I2S_DA_IN1..3			0.2	V _{SUP2}	
V _{I2SIH}	Input High Voltage		0.5			V _{SUP2}	
Z _{I2SI}	Input Impedance		5			pF	
I _{LEAKI2S}	Input Leakage Current		-1		1	μA	0 V < U _{INPUT} < DVSUP
V _{I2SOL}	I ² S Output Low Voltage	I2S_CL I2S_WS I2S_DA_OUT			0.4	V	I _{I2SOL} = 1 mA
V _{I2SOH}	I ² S Output High Voltage		V _{SUP2} - 0.3			V	I _{I2SOH} = -1 mA
f _{I2SOWS}	I ² S-Word Strobe Output Frequency	I2S_WS		48.0		kHz	
f _{I2SOCL}	I ² S-Clock Output Frequency	I2S_CL		1.536		MHz	
R _{I2S10/I2S20}	I ² S-Clock Output High/Low-Ratio		0.9	1.0	1.1		
Synchronous I²S Interface							
t _{s_I2S}	I ² S Input Setup Time before Rising Edge of Clock	I2S_DA_IN1/2 I2S_CL	12			ns	for details see Fig. 4-22 "I ² S timing diagram (synchronous interface)"
t _{h_I2S}	I ² S Input Hold Time after Rising Edge of Clock		40			ns	
t _{d_I2S}	I ² S Output Delay Time after Falling Edge of Clock	I2S_CL I2S_WS I2S_DA_OUT			28	ns	C _L =30 pF
f _{I2SWS}	I ² S-Word Strobe Input Frequency	I2S_WS		48.0		kHz	
f _{I2SCL}	I ² S-Clock Input Frequency	I2S_CL		1.536		MHz	
R _{I2SCL}	I ² S-Clock Input Ratio		0.9		1.1		
Asynchronous I²S Interface							
t _{s_I2S3}	I ² S3 Input Setup Time before Rising Edge of Clock	I2S_CL3 I2S_WS3 I2S_DA_IN3	4			ns	for details see Fig. 4-23 "I ² S timing diagram (asynchronous interface)"
t _{h_I2S3}	I ² S3 Input Hold Time after Rising Edge of Clock		40			ns	
f _{I2S3WS}	I ² S3-Word Strobe Input Frequency	I2S_WS3	5		50	kHz	
f _{I2S3CL}	I ² S3-Clock Input Frequency	I2S_CL3			3.2	MHz	
R _{I2S3CL}	I ² S3-Clock Input Ratio		0.9		1.1		



Data: MSB first, I²S synchronous master



Data: MSB first, I²S synchronous slave

Note:

- 1) I2S_DA_IN can be
 - I2S_DA_IN1,
 - I2S_DA_IN2, or
 - I2S_DA_IN2/3

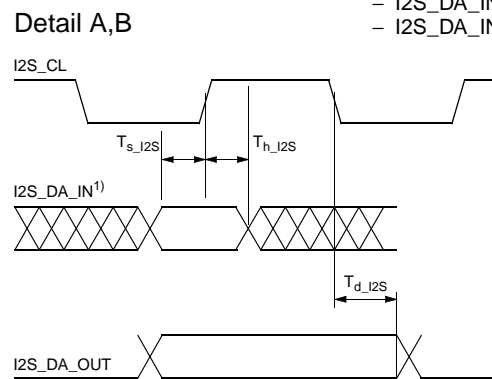
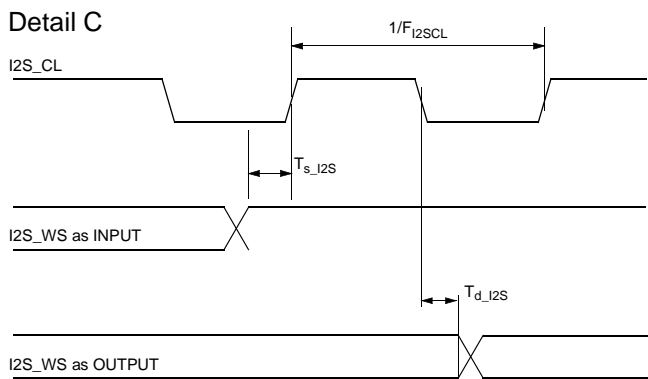


Fig. 4-22: I²S timing diagram (synchronous interface)

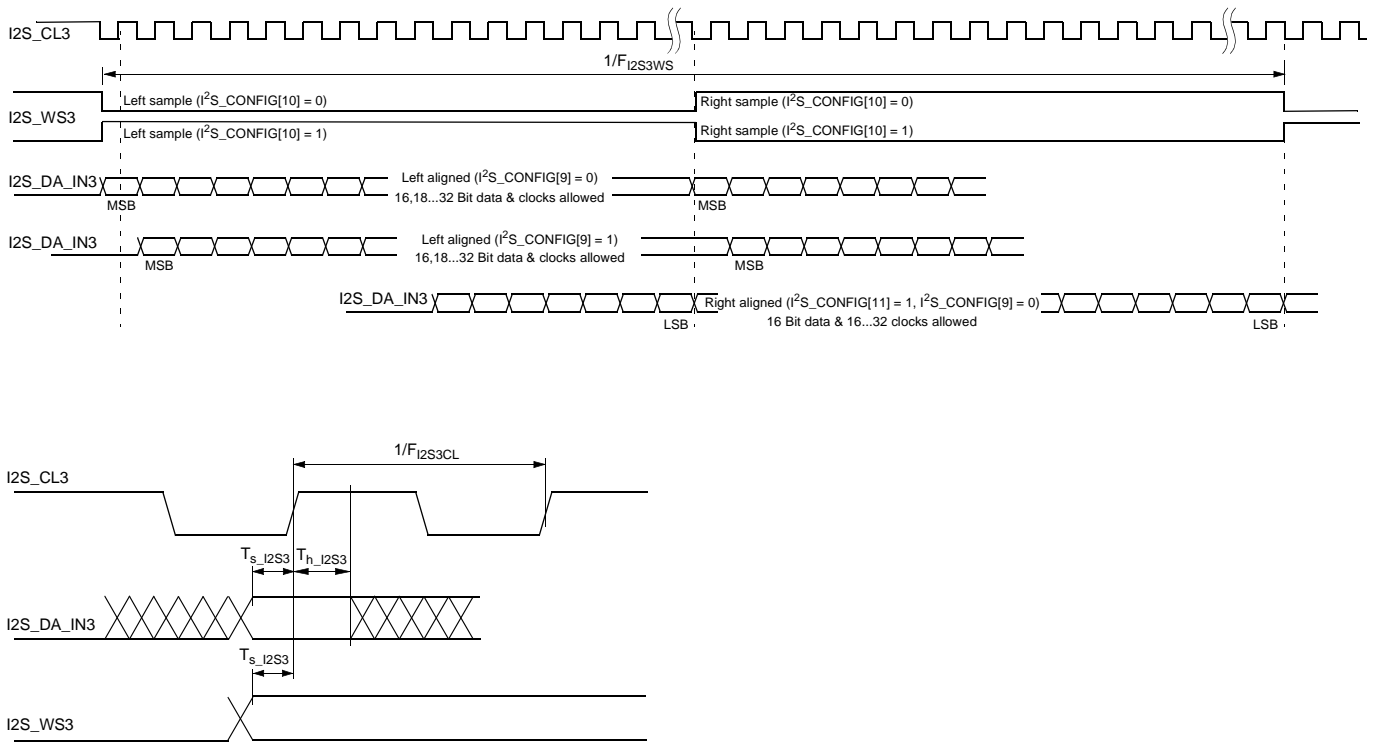


Fig. 4–23: I²S timing diagram (asynchronous interface)

4.6.3.6. Analog Baseband Inputs and Outputs, AGNDC

Symbol	Parameter	Pin Name	Min.	Typ.	Max.	Unit	Test Conditions
Analog Ground							
V_{AGNDC0}	AGNDC Open Circuit Voltage 8-V Operation: 5-V Operation:	AGNDC		3.77 2.49		V V	$R_{load} \geq 10 \text{ M}\Omega$
R_{outAGN}	AGNDC Output Resistance 8-V Operation: 5-V Operation:		70 47	125 83	180 120	k Ω k Ω	$3 \text{ V} \leq V_{AGNDC} \leq 4 \text{ V}$
Analog Input Resistance							
R_{inSC}	SCART Input Resistance from $T_A = 0$ to $70 \text{ }^\circ\text{C}$	$SCn_IN_s^{1)}$	25	40	58	k Ω	$f_{signal} = 1 \text{ kHz}, I = 0.05 \text{ mA}$
R_{inMONO}	MONO Input Resistance from $T_A = 0$ to $70 \text{ }^\circ\text{C}$	MONO_IN	15	24	35	k Ω	$f_{signal} = 1 \text{ kHz}, I = 0.1 \text{ mA}$
1) "n" means "1", "2", "3", or "4"; "s" means "L" or "R"							

Symbol	Parameter	Pin Name	Min.	Typ.	Max.	Unit	Test Conditions
Audio Analog-to-Digital-Converter							
V _{AICL}	Effective Analog Input Clipping Level for Analog-to-Digital-Conversion (8-V Operation)	SCn_IN_s, ¹⁾ MONO_IN	2.00		2.25	V _{RMS}	f _{signal} = 1 kHz
	Effective Analog Input Clipping Level for Analog-to-Digital-Conversion (5-V Operation)		1.13		1.51	V _{RMS}	
SCART Outputs							
R _{outSC}	SCART Output Resistance at T _j = 27 °C from T _A = 0 to 70 °C	SCn_OUT_s ¹⁾	200 200	330	460 500	Ω Ω	f _{signal} = 1 kHz, I = 0.1 mA
dV _{OUTSC}	Deviation of DC-Level at SCART Output from AGNDC Voltage		-70		+70	mV	
A _{SCtoSC}	Gain from Analog Input to SCART Output	SCn_IN_s, ¹⁾ MONO_IN	-1.0		+0.5	dB	f _{signal} = 1 kHz
f _{rSCtoSC}	Frequency Response from Analog Input to SCART Output (0 to 20000 Hz)	→ SCn_OUT_s ¹⁾	-0.5		+0.5	dB	with resp. to 1 kHz
V _{outSC}	Effective Signal Level at SCART-Output during full-scale Digital Input Signal from I ² S (8-V Operation)	SCn_OUT_s ¹⁾	1.8	1.9	2.0	V _{RMS}	f _{signal} = 1 kHz
	Effective Signal Level at SCART-Output during full-scale Digital Input Signal from I ² S (5-V Operation)		1.17	1.27	1.37	V _{RMS}	
Main and Aux Outputs							
R _{outMA}	Main/Aux Output Resistance at T _j = 27 °C from T _A = 0 to 70 °C	DACp_s ¹⁾	2.1 2.1	3.3	4.6 5.0	kΩ kΩ	f _{signal} = 1 kHz, I = 0.1 mA
V _{outDCMA}	DC-Level at Main/Aux-Output for Analog Volume at 0 dB for Analog Volume at -30 dB (8-V Operation)		1.80	2.04 61	2.28	V mV	
	DC-Level at Main/Aux-Output for Analog Volume at 0 dB for Analog Volume at -30 dB (5-V Operation)		1.12	1.36 40	1.60	V mV	
V _{outMA}	Effective Signal Level at Main/Aux-Output during full-scale Digital Input Signal from I ² S for Analog Volume at 0 dB (8-V Operation)		1.23	1.37	1.51	V _{RMS}	
	Effective Signal Level at Main/Aux-Output during full-scale Digital Input Signal from I ² S for Analog Volume at 0 dB (5-V Operation)	0.76	0.90	1.04	V _{RMS}		
1) "n" means "1", "2", "3", or "4"; "s" means "L" or "R"; "p" means "M" or "A"							

4.6.3.7. Sound IF Inputs

Symbol	Parameter	Pin Name	Min.	Typ.	Max.	Unit	Test Conditions
R _{IFIN}	Input Impedance	ANA_IN1+ ANA_IN2+ ANA_IN-	1.5 6.8	2 9.1	2.5 11.4	kΩ kΩ	Gain AGC = 20 dB Gain AGC = 3 dB
DC _{VREFTOP}	DC Voltage at VREFTOP		2.45	2.65	2.75	V	
DC _{ANA_IN}	DC Voltage on IF Inputs		1.3	1.5	1.7	V	
XTALK _{IF}	Crosstalk Attenuation		40			dB	f _{signal} = 1 MHz Input Level = -2 dB
BW _{IF}	3 dB Bandwidth		10			MHz	
AGC	AGC Step Width			0.85		dB	

4.6.3.8. Power Supply Rejection

Symbol	Parameter	Pin Name	Min.	Typ.	Max.	Unit	Test Conditions
PSRR: Rejection of Noise on AHVSUP at 1 kHz							
PSRR	AGNDC	AGNDC		80		dB	
	From Analog Input to I ² S Output	MONO_IN _n SCn_IN_s ¹⁾		70		dB	
	From Analog Input to SCART Output	MONO_IN _n SCn_IN_s ¹⁾ SCn_OUT_s ¹⁾		70		dB	
	From I ² S Input to SCART Output	SCn_OUT_s ¹⁾		60		dB	
	From I ² S Input to Main/Aux Output	DACp_s ¹⁾		80		dB	
1) "n" means "1", "2", "3", or "4"; "s" means "L" or "R"; "p" means "M" or "A"							

4.6.3.9. Analog Performance

Symbol	Parameter	Pin Name	Min.	Typ.	Max.	Unit	Test Conditions
Specifications for 8-V Operation							
SNR	Signal-to-Noise Ratio						
	from Analog Input to I ² S Output	MONO_IN, SCn_IN_s ¹⁾	90	93		dB	Input Level = -20 dB with resp. to V _{AICL} , f _{sig} = 1 kHz, A-weighted 20 Hz...20 kHz
	from Analog Input to SCART Output	MONO_IN, SCn_IN_s ¹⁾ → SCn_OUT_s ¹⁾	93	96		dB	
	from I ² S Input to SCART Output	SCn_OUT_s ¹⁾	90	93		dB	
	from I ² S Input to Main/Aux-Output	DACp_s ¹⁾	90	93		dB	
Total Harmonic Distortion							
THD	from Analog Input to I ² S Output	MONO_IN, SCn_IN_s ¹⁾		0.01	0.03	%	Input Level = -3 dBr with resp. to V _{AICL} , f _{sig} = 1 kHz, unweighted 20 Hz...20 kHz
	from Analog Input to SCART Output	MONO_IN, SCn_IN_s → SCn_OUT_s ¹⁾		0.01	0.03	%	
	from I ² S Input to SCART Output	SCn_OUT_s ¹⁾		0.01	0.03	%	
	from I ² S Input to Main or Aux Output	DACA_s, DACM_s ¹⁾		0.01	0.03	%	
	1) "n" means "1", "2", "3", or "4"; "s" means "L" or "R"; "p" means "M" or "A"						

Symbol	Parameter	Pin Name	Min.	Typ.	Max.	Unit	Test Conditions
Specifications for 5-V Operation							
SNR	Signal-to-Noise Ratio						
	from Analog Input to I ² S Output	MONO_IN, SCn_IN_s ¹⁾	87	90		dB	Input Level = -20 dB with resp. to V _{AICL} , f _{sig} = 1 kHz, A-weighted 20 Hz...20 kHz
	from Analog Input to SCART Output	MONO_IN, SCn_IN_s ¹⁾ → SCn_OUT_s ¹⁾	90	93		dB	Input Level = -20 dB, f _{sig} = 1 kHz, A-weighted 20 Hz...20 kHz Volume = 0 dB
	from I ² S Input to SCART Output	SCn_OUT_s ¹⁾	87	90		dB	
	from I ² S Input to Main/Aux-Output for Analog Volume at 0 dB for Analog Volume at -30 dB	DACp_s ¹⁾	87 75	90 80		dB dB	
THD	Total Harmonic Distortion						
	from Analog Input to I ² S Output	MONO_IN, SCn_IN_s ¹⁾		0.03	0.1	%	Input Level = -3 dBr with resp. to V _{AICL} , f _{sig} = 1 kHz, unweighted 20 Hz...20 kHz
	from Analog Input to SCART Output	MONO_IN, SCn_IN_s → SCn_OUT_s ¹⁾			0.1	%	Input Level = -3 dBr, f _{sig} = 1 kHz, unweighted 20 Hz...20 kHz
	from I ² S Input to SCART Output	SCn_OUT_s ¹⁾			0.1	%	
	from I ² S Input to Main or Aux Output	DACA_s, DACM_s ¹⁾			0.1	%	
1) "n" means "1", "2", "3", or "4"; "s" means "L" or "R"; "p" means "M" or "A"							

Symbol	Parameter	Pin Name	Min.	Typ.	Max.	Unit	Test Conditions					
Crosstalk Specifications for 8-V and 5-V Operation												
XTALK	Crosstalk Attenuation						Input Level = -3 dB, f _{sig} = 1 kHz, unused analog inputs connected to ground by Z < 1 kΩ					
	between left and right channel within SCART Input/Output pair (L→R, R→L) SCn_IN → SCn_OUT ¹⁾		80			dB	unweighted 20 Hz...20 kHz					
	SC1_IN or SC2_IN → I ² S Output		80			dB						
	SC3_IN → I ² S Output		80			dB						
	I ² S Input → SCn_OUT ¹⁾		80			dB						
	between left and right channel within Main or Aux Output pair I ² S Input → DACp ¹⁾		75			dB	unweighted 20 Hz...20 kHz					
between SCART Input/Output pairs ¹⁾ D = disturbing program O = observed program D: MONO/SCn_IN → SCn_OUT O: MONO/SCn_IN → SCn_OUT ¹⁾ D: MONO/SCn_IN → SCn_OUT or unsel. O: MONO/SCn_IN → I ² S Output D: MONO/SCn_IN → SCn_OUT O: I ² S Input → SCn_OUT ¹⁾ D: MONO/SCn_IN → unselected O: I ² S Input → SC1_OUT ¹⁾		100			dB	(unweighted 20 Hz...20 kHz) same signal source on left and right disturbing channel, effect on each observed output channel						
		95			dB							
		100			dB							
		100			dB							
	Crosstalk between Main and Aux Output pairs I ² S Input DSP → DACp ¹⁾		90			dB	(unweighted 20 Hz...20 kHz) same signal source on left and right disturbing channel, effect on each observed output channel					
XTALK	Crosstalk from Main or Aux Output to SCART Output and vice versa D = disturbing program O = observed program D: MONO/SCn_IN/DSP → SCn_OUT O: I ² S Input → DACp ¹⁾ D: MONO/SCn_IN/DSP → SCn_OUT O: I ² S Input → DACp ¹⁾ D: I ² S Input → DACp O: MONO/SCn_IN → SCn_OUT ¹⁾ D: I ² S Input → DACM O: I ² S Input → SCn_OUT ¹⁾							80			dB	(unweighted 20 Hz...20 kHz) same signal source on left and right disturbing channel, effect on each observed output channel SCART output load resistance 10 kΩ SCART output load resistance 30 kΩ
			85			dB						
			95			dB						
			95			dB						
			95			dB						
¹⁾ "n" means "1", "2", "3", or "4"; "s" means "L" or "R"; "p" means "M" or "A"												

4.6.3.10. Sound Standard Dependent Characteristics

Symbol	Parameter	Pin Name	Min.	Typ.	Max.	Unit	Test Conditions
NICAM Characteristics (MSP Standard Code = 8)							
$dV_{NICAMOUT}$	Tolerance of Output Voltage of NICAM Baseband Signal	DACp_s SCn_OUT_s ¹⁾	-1.5		+1.5	dB	2.12 kHz, Modulator input level = 0 dBref
S/N_{NICAM}	S/N of NICAM Baseband Signal		72			dB	NICAM: -6 dB, 1 kHz, RMS unweighted 0 to 15 kHz, Vol = 9 dB NIC_Presc = 7Fh Output level 1 V_{RMS} at DACp_s
THD_{NICAM}	Total Harmonic Distortion + Noise of NICAM Baseband Signal				0.1	%	2.12 kHz, Modulator input level = 0 dBref
BER_{NICAM}	NICAM: Bit Error Rate				1	10^{-7}	FM+NICAM, norm conditions
fR_{NICAM}	NICAM Frequency Response, 20...15000 Hz		-1.0		+1.0	dB	Modulator input level = -12 dB dBref; RMS
$XTALK_{NICAM}$	NICAM Crosstalk Attenuation (Dual)		80			dB	
SEP_{NICAM}	NICAM Channel Separation (Stereo)		80			dB	
FM Characteristics (MSP Standard Code = 3)							
dV_{FMOUT}	Tolerance of Output Voltage of FM Demodulated Signal	DACp_s, SCn_OUT_s ¹⁾	-1.5		+1.5	dB	1 FM-carrier, 50 μ s, 1 kHz, 40 kHz deviation; RMS
S/N_{FM}	S/N of FM Demodulated Signal		73			dB	1 FM-carrier 5.5 MHz, 50 μ s, 1 kHz, 40 kHz deviation; RMS, unweighted 0 to 15 kHz (for S/N); full input range, FM-Prescale = 46 h, Vol = 0 dB → Output Level 1 V_{RMS} at DACp_s
THD_{FM}	Total Harmonic Distortion + Noise of FM Demodulated Signal				0.1	%	
fR_{FM}	FM Frequency Responses, 20...15000 Hz		-1.0		+1.0	dB	1 FM-carrier 5.5 MHz, 50 μ s, Modulator input level = -14.6 dBref; RMS
$XTALK_{FM}$	FM Crosstalk Attenuation (Dual)		80			dB	2 FM-carriers 5.5/5.74 MHz, 50 μ s, 1 kHz, 40 kHz deviation; Bandpass 1 kHz
SEP_{FM}	FM Channel Separation (Stereo)		50			dB	2 FM-carriers 5.5/5.74 MHz, 50 μ s, 1 kHz, 40 kHz deviation; RMS
<p>1) "n" means "1", "2", "3", or "4"; "s" means "L" or "R"; "p" means "M" or "A"</p> <p>2) EIM refers to 75-μs Equivalent Input Modulation. It is defined as the audio-signal level which results in a stated percentage modulation, when the DBX encoding process is replaced by a 75-μs preemphasis network.</p>							

Symbol	Parameter	Pin Name	Min.	Typ.	Max.	Unit	Test Conditions	
AM Characteristics (MSP Standard Code = 9)								
$S/N_{AM(1)}$	S/N of AM Demodulated Signal measurement condition: RMS/Flat	DACp_s, SCn_OUT_s ¹⁾	55			dB	SIF level: 0.1–0.8 V _{pp} AM-carrier 54% at 6.5 MHz Vol = 0 dB, FM/AM prescaler set for output = 0.5 V _{RMS} at Main out; Standard Code = 09 _{hex} no video/chrominance components	
$S/N_{AM(2)}$	S/N of AM Demodulated Signal measurement condition: QP/CCIR		45			dB		
THD _{AM}	Total Harmonic Distortion + Noise of AM Demodulated Signal				0.6	%		
f _{RAM}	AM Frequency Response 50 Hz... 12 kHz		-2.5		+1.0	dB		
BTSC Characteristics (MSP Standard Code = 20_{hex}, 21_{hex})								
S/N_{BTSC}	S/N of BTSC Stereo Signal	DACp_s, SCn_OUT_s ¹⁾	68			dB	1 kHz L or R or SAP, 100% modulation, 75 μs deem- phasis, RMS unweighted 0 to 15 kHz	
	S/N of BTSC-SAP Signal		57			dB		
THD _{BTSC}	THD+N of BTSC Stereo Signal THD+N of BTSC SAP Signal				0.1	%		1 kHz L or R or SAP, 100% 75 μs EIM ²⁾ , DBX NR, RMS unweighted 0 to 15 kHz
					0.5	%		
f _{RBTSC}	Frequency Response of BTSC Stereo, 50 Hz...12 kHz		-0.5		0.5	dB		L or R or SAP, 1%...66% EIM ²⁾ , DBX NR
	Frequency Response of BTSC SAP, 50 Hz...9 kHz		-1.0		0.6	dB		
XTALK _{BTSC}	Stereo → SAP SAP → Stereo		76			dB		1 kHz L or R or SAP, 100% modulation, 75 μs deem- phasis, Bandpass 1 kHz
		80			dB			
Sep _{BTSC}	Stereo Separation 50 Hz...10 kHz 50 Hz...12 kHz	35			dB	L or R 1%...66% EIM ²⁾ , DBX NR		
		30			dB			
FM _{ThrPilot}	Pilot deviation threshold Stereo off → on Stereo on → off	ANA_IN1+, ANA_IN2+	3.2		3.5	kHz	4.5 MHz carrier modulated with f _p =15.743 kHz SIF level=100mV _{pp} indication: STATUS Bit[6]	
			1.2		1.5	kHz		
f _{Pilot}	Pilot Frequency Range	ANA_IN1+ ANA_IN2+	15.563		15.843	kHz	standard BTSC stereo sig- nal, sound carrier only	
<p>1) "n" means "1", "2", "3", or "4"; "s" means "L" or "R"; "p" means "M" or "A"</p> <p>2) EIM refers to 75-μs Equivalent Input Modulation. It is defined as the audio-signal level which results in a stated percentage modulation, when the DBX encoding process is replaced by a 75-μs preemphasis network.</p>								

Symbol	Parameter	Pin Name	Min.	Typ.	Max.	Unit	Test Conditions	
BTSC Characteristics (MSP Standard Code = 20_{hex}, 21_{hex}) with a minimum IF input signal level of 70 mVpp (measured without any video/chroma signal components)								
S/N _{BTSC}	S/N of BTSC Stereo Signal	DACp _s , SCn_OUT _s ¹⁾	64			dB	1 kHz L or R or SAP, 100% modulation, 75 μs deemphasis, RMS unweighted 0 to 15 kHz	
	S/N of BTSC-SAP Signal		55			dB		
THD _{BTSC}	THD+N of BTSC Stereo Signal					0.15	%	1 kHz L or R or SAP, 100% modulation, 75 μs EIM ²⁾ , DBX NR, RMS unweighted 0 to 15 kHz
	THD+N of BTSC SAP Signal					0.8	%	
f _R _{BTSC}	Frequency Response of BTSC Stereo, 50 Hz...12 kHz			-0.5		0.5	dB	L or R or SAP, 1%...66% EIM ²⁾ , DBX NR
	Frequency Response of BTSC-SAP, 50 Hz...9 kHz			-1.0		0.6	dB	
XTALK _{BTSC}	Stereo → SAP		75			dB	1 kHz L or R or SAP, 100% modulation, 75 μs deemphasis, Bandpass 1 kHz	
	SAP → Stereo		75			dB		
Sep _{BTSC}	Stereo Separation 50 Hz...10 kHz 50 Hz...12 kHz		35			dB	L or R 1%...66% EIM ²⁾ , DBX NR	
			30			dB		
EIA-J Characteristics (MSP Standard Code = 30_{hex})								
S/N _{EIAJ}	S/N of EIA-J Stereo Signal	DACp _s , SCn_OUT _s ¹⁾	60			dB	1 kHz L or R, 100% modulation, 75 μs deemphasis, RMS unweighted 0 to 15 kHz	
	S/N of EIAJ Sub-Channel		60			dB		
THD _{EIAJ}	THD+N of EIA-J Stereo Signal					0.2	%	100% modulation, 75 μs deemphasis
	THD+N of EIA-J Sub-Channel					0.3	%	
f _R _{EIAJ}	Frequency Response of EIA-J Stereo, 50 Hz...12 kHz			-0.5		0.5	dB	100% modulation, 75 μs deemphasis
	Frequency Response of EIA-J Sub-Channel, 50 Hz...12 kHz			-1.0		0.5	dB	
XTALK _{EIAJ}	Main → SUB		66			dB	1 kHz L or R, 100% modulation, 75 μs deemphasis, Bandpass 1 kHz	
	Sub → Main		80			dB		
SEP _{EIAJ}	Stereo Separation 50 Hz...5 kHz 50 Hz...10 kHz		35			dB	EIA-J Stereo Signal, L or R 100% modulation	
			28			dB		
FM-Radio Characteristics (MSP Standard Code = 40_{hex})								
S/N _{UKW}	S/N of FM-Radio Stereo Signal	DACp _s , SCn_OUT _s ¹⁾	70			dB	1 kHz L or R, 100% modulation, 75 μs deemphasis, RMS unweighted 0 to 15 kHz	
THD _{UKW}	THD+N of FM-Radio Stereo Signal					0.1		%
f _R _{UKW}	Frequency Response of FM-Radio Stereo 50 Hz...15 kHz			-1.0		0.5	dB	L or R, 1%...100% modulation, 75 μs deemphasis
Sep _{UKW}	Stereo Separation 50 Hz...15 kHz			45			dB	
f _{Pilot}	Pilot Frequency Range		ANA_IN1+ ANA_IN2+	18.844		19.125	kHz	standard FM radio stereo signal
<p>1) "n" means "1", "2", "3", or "4"; "s" means "L" or "R"; "p" means "M" or "A"</p> <p>2) EIM refers to 75-μs Equivalent Input Modulation. It is defined as the audio-signal level which results in a stated percentage modulation, when the DBX encoding process is replaced by a 75-μs preemphasis network.</p>								

5. Appendix A: Overview of TV-Sound Standards

5.1. NICAM 728

Table 5–1: Summary of NICAM 728 sound modulation parameters

Specification	I	B/G	L		D/K	
Carrier frequency of digital sound	6.552 MHz	5.85 MHz	5.85 MHz		5.85 MHz	
Transmission rate	728 kbit/s					
Type of modulation	Differentially encoded quadrature phase shift keying (DQPSK)					
Spectrum shaping Roll-off factor	by means of Roll-off filters					
	1.0	0.4	0.4		0.4	
Carrier frequency of analog sound component	6.0 MHz FM mono	5.5 MHz FM mono	6.5 MHz AM mono terrestrial cable		6.5 MHz FM mono	
Power ratio between vision carrier and analog sound carrier	10 dB	13 dB	10 dB	16 dB	13 dB	
Power ratio between analog and modulated digital sound carrier	10 dB	7 dB	17 dB	11 dB	China/ Hungary	Poland
					12 dB	7 dB

Table 5–2: Summary of NICAM 728 sound coding characteristics

Characteristics	Values
Audio sampling frequency	32 kHz
Number of channels	2
Initial resolution	14 bit/sample
Companding characteristics	near instantaneous, with compression to 10 bits/sample in 32-samples (1 ms) blocks
Coding for compressed samples	2's complement
Preemphasis	CCITT Recommendation J.17 (6.5 dB attenuation at 800 Hz)
Audio overload level	+12 dBm measured at the unity gain frequency of the preemphasis network (2 kHz)

5.2. A2-Systems

Table 5–3: Key parameters for A2 Systems of Standards B/G, D/K, and M

Characteristics	Sound Carrier FM1			Sound Carrier FM2		
	B/G	D/K	M	B/G	D/K	M
TV-Sound Standard						
Carrier frequency in MHz	5.5	6.5	4.5	5.7421875	6.2578125 6.7421875	4.724212
Vision/sound power difference	13 dB			20 dB		
Sound bandwidth	40 Hz to 15 kHz					
Preemphasis	50 μ s		75 μ s	50 μ s		75 μ s
Frequency deviation (nom/max)	$\pm 27/\pm 50$ kHz		$\pm 17/\pm 25$ kHz	$\pm 27/\pm 50$ kHz		$\pm 15/\pm 25$ kHz
Transmission Modes						
Mono transmission	mono			mono		
Stereo transmission	(L+R)/2		(L+R)/2	R	(L–R)/2	
Dual sound transmission	language A			language B		
Identification of Transmission Mode						
Pilot carrier frequency				54.6875 kHz	55.0699 kHz	
Max. deviation portion				± 2.5 kHz		
Type of modulation / modulation depth				AM / 50%		
Modulation frequency				mono: unmodulated stereo: 117.5 Hz dual: 274.1 Hz	149.9 Hz 276.0 Hz	

5.3. BTSC-Sound System

Table 5–4: Key parameters for BTSC-Sound Systems

	Aural Carrier	BTSC-MPX-Components				
		(L+R)	Pilot	(L–R)	SAP	Prof. Ch.
Carrier frequency ($f_h = 15.734$ kHz)	4.5 MHz	Baseband	f_h	$2 f_h$	$5 f_h$	$6.5 f_h$
Sound bandwidth in kHz		0.05 - 15		0.05 - 15	0.05 - 12	0.05 - 3.4
Preemphasis		75 μ s		DBX	DBX	150 μ s
Max. deviation to Aural Carrier	73 kHz (total)	25 kHz ¹⁾	5 kHz	50 kHz ¹⁾	15 kHz	3 kHz
Max. Freq. Deviation of Subcarrier Modulation Type				AM	10 kHz FM	3 kHz FM
1) Sum does not exceed 50 kHz due to interleaving effects						

5.4. Japanese FM Stereo System (EIA-J)

Table 5–5: Key parameters for Japanese FM-Stereo Sound System EIA-J

	Aural Carrier FM	EIA-J-MPX-Components		
		(L+R)	(L–R)	Identification
Carrier frequency ($f_h = 15.734$ kHz)	4.5 MHz	Baseband	$2 f_h$	$3.5 f_h$
Sound bandwidth		0.05 - 15 kHz	0.05 - 15 kHz	–
Preemphasis		75 μ s	75 μ s	none
Max. deviation portion to Aural Carrier	47 kHz	25 kHz	20 kHz	2 kHz
Max. Freq. Deviation of Subcarrier Modulation Type			10 kHz FM	60% AM
Transmitter-sided delay		20 μ s	0 μ s	0 μ s
Mono transmission		L+R	–	unmodulated
Stereo transmission		L+R	L–R	982.5 Hz
Bilingual transmission		Language A	Language B	922.5 Hz

5.5. FM Satellite Sound

Table 5–6: Key parameters for FM Satellite Sound

Carrier Frequency	Maximum FM Deviation	Sound Mode	Bandwidth	Deemphasis
6.5 MHz	85 kHz	Mono	15 kHz	50 μ s
7.02/7.20 MHz	50 kHz	Mono/Stereo/Bilingual	15 kHz	adaptive
7.38/7.56 MHz	50 kHz	Mono/Stereo/Bilingual	15 kHz	adaptive
7.74/7.92 MHz	50 kHz	Mono/Stereo/Bilingual	15 kHz	adaptive

5.6. FM-Stereo Radio

Table 5–7: Key parameters for FM-Stereo Radio Systems

	Aural Carrier	FM-Radio-MPX-Components			
		(L+R)	Pilot	(L–R)	RDS/ARI
Carrier frequency ($f_p = 19$ kHz)	10.7 MHz	Baseband	f_p	$2 f_p$	$3 f_p$
Sound bandwidth in kHz		0.05 - 15		0.05 - 15	
Preemphasis: – USA – Europe		75 μ s 50 μ s		75 μ s 50 μ s	
Max. deviation to Aural Carrier	75 kHz (100%)	90% ¹⁾	10%	90% ¹⁾	5%
¹⁾ Sum does not exceed 90% due to interleaving effects					

6. Appendix B: Manual Mode

To adapt the modes of the STANDARD SELECT register to individual requirements, the MSP 44x8G offers a Manual Mode, which provides sophisticated programming of the MSP 44x8G.

The Manual Mode can be used only in those cases, where user specific requirements concerning detection, identification, or carrier positioning have to be met.

After the setting of the STANDARD SELECT register, the MSP 44x8G is set up for optimal behavior. **Therefore, it is not recommended to use the Manual mode.**

6.1. Demodulator Write and Read Registers for Manual Mode

In case of Automatic Sound Select (MODUS[0]=1), any modifications of all DCO registers listed in Table 6–1 are ignored.

Table 6–1: Demodulator Write Registers; Subaddress: 10_{hex}; these registers are not readable!

Demodulator Write Registers	Address (hex)	MSP-Version	Description	Reset Mode	Page
AUTO_FM/AM	00 21	4418, 4458 ¹⁾	1. MODUS[0]=1 (Automatic Sound Select): Switching Level threshold of Automatic Switching between NICAM and FM/AM in case of bad NICAM reception 2. MODUS[0]=0 (Manual Mode): Activation and configuration of Automatic Switching between NICAM and FM/AM in case of bad NICAM reception	00 00 _{hex}	74
A2_Threshold	00 22		A2 Stereo Identification Threshold	00 19 _{hex}	
CM_Threshold	00 24		Carrier-Mute Threshold	00 2A _{hex}	
DCO1_LO DCO1_HI DCO2_LO DCO2_HI	00 93 00 9B 00 A3 00 AB		Note: Modifications are ignored for Automatic Sound Select = on (MODUS[0]=1) Increment channel 1 Low Part Increment channel 1 High Part Increment channel 2 Low Part Increment channel 2 High Part	00 00 _{hex}	77

¹⁾ not in BTSC, EIA-J, and FM-Radio mode

Table 6–2: Demodulator Read Registers; Subaddress: 11_{hex}; these registers are not writable!

Demodulator Read Registers	Address (hex)	MSP-Version	Description	Page
C_AD_BITS	00 23	4410, 4450	NICAM-Sync bit, NICAM-C-Bits, and bit[2:0] of additional data bits	78
ADD_BITS	00 38		NICAM: bit[10:3] of additional data bits	78
CIB_BITS	00 3E		NICAM: CIB1 and CIB2 control bits	78
ERROR_RATE	00 57		NICAM error rate, updated with 182 ms	78

6.2. DSP Write and Read Registers for Manual Mode

Table 6–3: DSP-Write Registers; Subaddress: 12_{hex}, all registers are readable as well

Write Register	Address (hex)	Bits	Operational Modes and Adjustable Range	Reset Mode	Page
Additional Channel Matrix Modes	00 08 00 09 00 0A 00 41 00 0B 00 0C	[7:0]	[SUM/DIFF, AB_XCHANGE, PHASE_CHANGE_B, PHASE_CHANGE_A, A_ONLY, B_ONLY]	00 _{hex}	79
FM Fixed Deemphasis	00 0F	[15:8]	[OFF, 50 μs, 75 μs]	OFF	79
FM Adaptive Deemphasis		[7:0]	[OFF, WP1]	OFF	79
Identification Mode	00 15	[7:0]	[B/G, M]	B/G	79

Table 6–4: DSP Read Registers; Subaddress: 13_{hex}, all registers are not writable

Additional Read Registers	Address (hex)	Bits	Output Range	Page
Stereo detection register for A2 Stereo Systems	00 18	[15:8]	[80 _{hex} ... 7F _{hex}] 8 bit two's complement	80
DC level readout FM1/Ch2-L	00 1B	[15:0]	[8000 _{hex} ... 7FFF _{hex}] 16 bit two's complement	80
DC level readout FM2/Ch1-R	00 1C	[15:0]	[8000 _{hex} ... 7FFF _{hex}] 16 bit two's complement	80

6.3. Manual Mode:
Description of Demodulator Write Registers

6.3.1. Automatic Switching between NICAM and Analog Sound

In case of bad NICAM reception or loss of the NICAM-carrier, the MSP 44x8G offers an Automatic Switching (fall back) to the analog sound (FM/AM-Mono), without the necessity of the controller reading and evaluating any parameters. If a proper NICAM signal returns, switching back to this source is performed automatically as well. The feature evaluates the NICAM ERROR_RATE and switches, if necessary, all output channels which are assigned to the NICAM source, to the analog source, and vice versa.

An appropriate hysteresis algorithm avoids oscillating effects (see Fig. 6–1). STATUS[9] and C_AD_BITS[11] (Addr: 0023_{hex}) provide information about the actual NICAM-FM/AM-status.

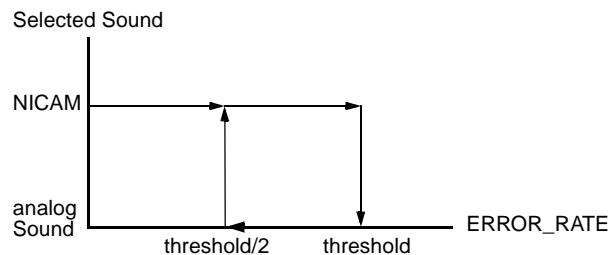


Fig. 6–1: Hysteresis for automatic switching

6.3.1.1. Function in Automatic Sound Select Mode

The Automatic Sound Select feature (MODUS[0]=1) includes the procedure mentioned above. By default, the internal ERROR_RATE threshold is set to 700_{dec}. i.e.:

- NICAM → analog sound if ERROR_RATE > 700
- analog sound → NICAM if ERROR_RATE < 700/2

The ERROR_RATE value of 700 corresponds to a BER of approximately $5.46 \cdot 10^{-3}/s$.

Individual configuration of the threshold can be done using Table 6–5, whereby the bits [0] and [11] of AUTO_FM are ignored. It is recommended to use the internal setting used by the standard selection.

The optimum NICAM sound can be assigned to the MSP output channels by selecting one of the “Stereo or A/B”, “Stereo or A”, or “Stereo or B” source channels.

6.3.1.2. Function in Manual Mode

If the manual mode (MODUS[0]=0) is required, the activation and configuration of the Automatic Switching feature has to be done as described in Table 6–5. Note, that the channel matrix of the corresponding output channels must be set according to the NICAM mode and need not to be changed in the FM/AM-fallback case.

Example:

Required threshold = 500: bits [10:1]=00 1111 1010

Table 6–5: Coding of Automatic NICAM/Analog Sound Switching;
Reset Status: Mode 0;
Automatic Sound Select is on (MODUS[0] = 1)

Mode	Description	AUTO_FM [11:0] Addr. = 00 21 _{hex}	ERROR_RATE- Threshold/dec	Source Select: Input at NICAM Path ¹⁾
1	Automatic Switching with internal threshold (Default, if Automatic Sound Select is on)	Bit[11] = ignored Bit[10:1] = 0 Bit[0] = ignored	700	NICAM or FM/AM, depending on ERROR_RATE
2	Automatic Switching with external threshold (Customizing of Automatic Sound Select)	Bit[11] = ignored Bit[10:1] = 25...1000 = threshold/2 Bit[0] = ignored	set by customer; recommended range: 50...2000	
¹⁾ The NICAM path may be assigned to “Stereo or A/B”, “Stereo or A”, or “Stereo or B” source channels (see Table 2–2 on page 11).				

Table 6–6: Coding of Automatic NICAM/Analog Sound Switching;
Reset Status: Mode 0;
Automatic Sound Select is off (MODUS[0] = 0)

Mode	Description	AUTO_FM [11:0] Addr. = 00 21 _{hex}	ERROR_RATE- Threshold/dec	Source Select: Input at NICAM Path
0	Forced NICAM (Automatic Switching disabled)	Bit[11] = 0 Bit[10:1] = 0 Bit[0] = 0	none	always NICAM; Mute in case of no NICAM available
1	Automatic Switching with internal threshold (Default, if Automatic Sound Select is on)	Bit[11] = 0 Bit[10:1] = 0 Bit[0] = 1	700	NICAM or FM/AM, depending on ERROR_RATE
2	Automatic Switching with external threshold (Customizing of Automatic Sound Select)	Bit[11] = 0 Bit[10:1] = 25...1000 = threshold/2 Bit[0] = 1	set by customer; recommended range: 50...2000	
3	Forced Analog Mono (Automatic Switching disabled)	Bit[11] = 1 Bit[10:1] = 0 Bit[0] = 1	none	always FM/AM

6.3.2. A2 Threshold

The threshold between Stereo/Bilingual and Mono Identification for the A2 Standard has been made programmable according to the user's preferences.

An internal hysteresis ensures robustness and stability.

Table 6–7: Write Register on I²C Subaddress 10_{hex}: A2 Threshold

Register Address	Function	Name
THRESHOLDS		
00 22 _{hex} (write)	<p>A2 THRESHOLD Register</p> <p>Defines threshold of all A2 and EIA_J standards for Stereo and Bilingual detection</p> <p>bit[11:0] 7F0_{hex} force Mono Identification</p> <p>... 190_{hex} default setting after reset</p> <p>... 0A0_{hex} minimum Threshold for stable detection</p> <p>recommended range: 0A_{hex}...3C_{hex}</p>	A2_THRESH

6.3.3. Carrier-Mute Threshold

The Carrier-Mute threshold has been made programmable according to the users preferences. An internal hysteresis ensures stable behavior.

Table 6–8: Write Register on I²C Subaddress 10_{hex}: Carrier-Mute Threshold

Register Address	Function	Name
THRESHOLDS		
00 24 _{hex} (write)	<p>Carrier-Mute THRESHOLD Register</p> <p>Defines threshold for the carrier mute feature</p> <p>bit[6:0] 00_{hex} Carrier-Mute always ON (both channels muted)</p> <p>... 2A_{hex} default setting after reset</p> <p>... FF_{hex} Carrier-Mute always OFF (both channels forced on)</p> <p>recommended range: 14_{hex}...50_{hex}</p>	CM_THRESH

6.3.4. DCO-Registers

Note: The use of this register is not recommended. It should be used only in cases where non-standard carrier frequencies have to be processed. Please note, that the usage of user specific demodulation frequencies is not possible in combination with the Automatic Sound Select (MODUS[0]=1).

When selecting a TV-sound standard by means of the STANDARD SELECT register, all frequency tuning is performed automatically.

If manual setting of the tuning frequency is required, a set of 24-bit registers determining the mixing frequencies of the quadrature mixers can be written manually into the MSP. In Table 6–9, examples for DCO register programming are listed. It is necessary to separate these registers into two categories: low part and high part. The formula for the calculation of the INCR values for any chosen IF frequency is as follows:

$$INCR_{dec} = \text{int} (f / f_s \cdot 2^{24})$$

with: int = integer function
 f = IF frequency in MHz
 f_s = sampling frequency (18.432 MHz)

Conversion of INCR into hex-format and separation of the 12-bit low and high parts lead to the required register values (DCO1_HI and _LO for MSP-Ch1, DCO2_HI and _LO for MSP-Ch2).

**6.4. Manual Mode:
 Description of Demodulator Read Registers**

Note: This register should be used only in cases where software compatibility to the MSP 44x0D is required. Using the STANDARD SELECTION register together with the STATUS register provides a more economic way to program the MSP 44x8G and to retrieve information from the MSP.

All registers except C_AD_BITS are 8 bits wide. They can be read out of the RAM of the MSP 44x8G.

All transmissions take place in 16-bit words. The valid 8-bit data are the 8 LSBs of the received data word.

If the Automatic Sound Select feature is not used, the NICAM or FM-identification parameters must be read and evaluated by the controller in order to enable appropriate switching of the channel select matrix of the baseband processing part. The FM-identification registers are described in Section 6.6.1. To handle the NICAM-sound and to observe the NICAM-quality, at least the registers C_AD_BITS and ERROR_RATE must be read and evaluated by the controller. Additional data bits and CIB bits, if supplied by the NICAM transmitter, can be obtained by reading the registers ADD_BITS and CIB_BITS.

Table 6–9: DCO registers for the MSP 44x8G; reset status: DCO_HI/LO = “00 00”

DCO1_LO 00 93 _{hex} , DCO1_HI 00 9B _{hex} ; DCO2_LO 00 A3 _{hex} , DCO2_HI 00 AB _{hex}					
IF-Freq. [MHz]	DCO_HI [hex]	DCO_LO [hex]	IF-Freq. [MHz]	DCO_HI [hex]	DCO_LO [hex]
4.5	03 E8	00 00			
5.04	04 60	00 00	5.76	05 00	00 00
5.5	04 C6	03 8E	5.85	05 14	00 00
5.58	04 D8	00 00	5.94	05 28	00 00
5.7421875	04 FC	00 AA			
6.0	05 35	05 55	6.6	05 BA	0A AA
6.2	05 61	0C 71	6.65	05 C5	0C 71
6.5	05 A4	07 1C	6.8	05 E7	01 C7
6.552	05 B0	00 00			
7.02	06 18	00 00	7.2	06 40	00 00
7.38	06 68	00 00	7.56	06 90	00 00

6.4.1. NICAM Mode Control/Additional Data Bits Register

NICAM operation mode control bits and A[2:0] of the additional data bits.

Format:

MSB		C_AD_BITS 00 23 _{hex}						LSB	
11	...	7	6	5	4	3	2	1	0
Auto_FM	...	A[2]	A[1]	A[0]	C4	C3	C2	C1	S

Important: “S” = Bit[0] indicates correct NICAM-synchronization (S = 1). If S = 0, the MSP 4418/4458G has not yet synchronized correctly to frame and sequence, or has lost synchronization. The remaining read registers are therefore not valid. The MSP mutes the NICAM output automatically and tries to synchronize again as long as any NICAM standard is selected by the STANDARD SELECT register.

The operation mode is coded by C4-C1 as shown in Table 6–10.

Table 6–10: NICAM operation modes as defined by the EBU NICAM 728 specification

C4	C3	C2	C1	Operation Mode
0	0	0	0	Stereo sound (NICAMA/B), independent mono sound (FM1)
0	0	0	1	Two independent mono signals (NICAMA, FM1)
0	0	1	0	Three independent mono channels (NICAMA, NICAMB, FM1)
0	0	1	1	Data transmission only; no audio
1	0	0	0	Stereo sound (NICAMA/B), FM1 carries same channel
1	0	0	1	One mono signal (NICAMA). FM1 carries same channel as NICAMA
1	0	1	0	Two independent mono channels (NICAMA, NICAMB). FM1 carries same channel as NICAMA
1	0	1	1	Data transmission only; no audio
x	1	x	x	Unimplemented sound coding option (not yet defined by EBU NICAM 728 specification)

AUTO_FM: monitor bit for the AUTO_FM Status:
 0: NICAM source is NICAM
 1: NICAM source is FM

Note: It is not necessary to read out and evaluate the C_AD_BITS. All evaluation is performed in the MSP and indicated in the STATUS register.

6.4.2. Additional Data Bits Register

Contains the remaining 8 of the 11 additional data bits. The additional data bits are not yet defined by the NICAM 728 system.

Format:

MSB		ADD_BITS 00 38 _{hex}						LSB	
7	6	5	4	3	2	1	0		
A[10]	A[9]	A[8]	A[7]	A[6]	A[5]	A[4]	A[3]		

6.4.3. CIB Bits Register

CIB bits 1 and 2 (see NICAM 728 specifications).

Format:

MSB		CIB_BITS 00 3E _{hex}						LSB	
7	6	5	4	3	2	1	0		
x	x	x	x	x	x	CIB1	CIB2		

6.4.4. NICAM Error Rate Register

ERROR_RATE	00 57 _{hex}
Error free	0000 _{hex}
maximum error rate	07FF _{hex}

Average error rate of the NICAM reception in a time interval of 182 ms, which should be close to 0. The initial and maximum value of ERROR_RATE is 2047. This value is also active if no NICAM-standard is selected. Since the value is achieved by filtering, a certain transition time (approx. 0.5 sec) is unavoidable. Acceptable audio may have error rates up to a value of 700_{dec}. Individual evaluation of this value by the controller and an appropriate threshold may define the fallback mode from NICAM to FM/AM-Mono in case of poor NICAM reception.

The bit error rate per second (BER) can be calculated by means of the following formula:

$$\text{BER} = \text{ERROR_RATE} * 12.3 * 10^{-6} / \text{s}$$

6.5. Manual Mode: Description of DSP Write Registers

6.5.1. Additional Channel Matrix Modes

Main Matrix	00 08 _{hex}	L
Aux Matrix	00 09 _{hex}	L
SCART1 Matrix	00 0A _{hex}	L
SCART2 Matrix	00 41 _{hex}	L
I ² S Matrix	00 0B _{hex}	L
Quasi-Peak Detector Matrix	00 0C _{hex}	L
SUM/DIFF	0100 0000	40 _{hex}
AB_XCHANGE	0101 0000	50 _{hex}
PHASE_CHANGE_B	0110 0000	60 _{hex}
PHASE_CHANGE_A	0111 0000	70 _{hex}
A_ONLY	1000 0000	80 _{hex}
B_ONLY	1001 0000	90 _{hex}

This table shows additional modes for the channel matrix registers.

The sum/difference mode can be used together with the quasi-peak detector to determine the sound material mode. If the difference signal on channel B (right) is near to zero, and the sum signal on channel A (left) is high, the incoming audio signal is mono. If there is a significant level on the difference signal, the incoming audio is stereo.

6.5.2. FM Fixed Deemphasis

FM Deemphasis	00 0F _{hex}	H
50 μ s	0000 0000 RESET	00 _{hex}
75 μ s	0000 0001	01 _{hex}
OFF	0011 1111	3F _{hex}

6.5.3. FM Adaptive Deemphasis

FM Adaptive Deemphasis WP1	00 0F _{hex}	L
OFF	0000 0000 RESET	00 _{hex}
WP1	0011 1111	3F _{hex}

Note: The Adaptive Deemphasis WP1 requires setting of fixed deemphasis to 75 μ s.

6.5.4. NICAM Deemphasis

A J17 Deemphasis is always applied to the NICAM signal. It is not switchable.

6.5.5. Identification Mode for A2 Stereo Systems

Identification Mode	00 15 _{hex}	L
Standard B/G (German Stereo)	0000 0000 RESET	00 _{hex}
Standard M (Korean Stereo)	0000 0001	01 _{hex}
Reset of Ident-Filter	0011 1111	3F _{hex}

To shorten the response time of the identification algorithm after a program change between two FM-Stereo capable programs, the reset of the ident-filter can be applied.

Sequence:

1. Program change
2. Reset ident-filter
3. Set identification mode back to standard B/G or M
4. Read stereo detection register

**6.6. Manual Mode:
Description of DSP Read Registers**

All readable registers are 16-bit wide. Transmissions via I²C bus have to take place in 16-bit words. Some of the defined 16-bit words are divided into low and high byte, thus holding two different control entities.

These registers are not writable.

**6.6.1. Stereo Detection Register
for A2 Stereo Systems**

Stereo Detection Register	00 18_{hex}	H
Stereo Mode	Reading (two's complement)	
MONO	near zero	
STEREO	positive value (ideal reception: 7F _{hex})	
BILINGUAL	negative value (ideal reception: 80 _{hex})	

Note: It is not necessary to read out and evaluate the A2 identification level. All evaluation is performed in the MSP and indicated in the STATUS register.

6.6.2. DC Level Register

DC Level Readout FM1 (MSP-Ch2)	00 1B_{hex}	H+L
DC Level Readout FM2 (MSP-Ch1)	00 1C_{hex}	H+L
DC Level	[8000 _{hex} ... 7FFF _{hex}] values are 16 bit two's complement	

The DC level register measures the DC component of the incoming FM signals (FM1 and FM2). This can be used for seek functions in satellite receivers and for IF FM frequencies fine tuning. If the DCO frequency is lower than the actual carrier frequency, the resulting DC level will be positive, and vice versa. In the audio signal the DC content is suppressed. The time constant τ , defining the transition time of the DC Level Register, is approximately 28 ms.

6.7. Demodulator Source Channels in Manual Mode

6.7.1. Terrestrial Sound Standards

Table 6–11 shows the source channel assignment of the demodulated signals in case of manual mode for all terrestrial sound standards. See Table 2–2 for the assignment in the Automatic Sound Select mode. In manual mode for terrestrial sound standards, only two demodulator sources are defined.

6.7.2. SAT Sound Standards

Table 6–12 shows the source channel assignment of the demodulated signals for SAT sound standards.

Table 6–11: Manual Sound Select Mode for Terrestrial Sound Standards

				Source Channels of Sound Select Block	
Broadcasted Sound Standard	Selected MSP Standard Code	Broadcasted Sound Mode	FM Matrix	FM/AM (use 0 for channel select)	Stereo or A/B (use 1 for channel select)
B/G-FM D/K-FM M-Korea M-Japan	03 04, 05 02 30	MONO	Sound A Mono	Mono	Mono
		STEREO	German Stereo Korean Stereo	Stereo	Stereo
		BILINGUAL, Languages A and B	No Matrix	Left = A Right = B	Left = A Right = B
B/G-NICAM L-NICAM I-NICAM D/K-NICAM D/K-NICAM (with high deviation FM)	08 09 0A 0B 0C	NICAM not available or NICAM error rate too high	Sound A Mono	analog Mono	no sound with AUTO_FM: analog Mono
		MONO	Sound A Mono	analog Mono	NICAM Mono
		STEREO	Sound A Mono	analog Mono	NICAM Stereo
		BILINGUAL, Languages A and B	Sound A Mono	analog Mono	Left = NICAM A Right = NICAM B
BTSC	20	MONO	Sound A Mono	Mono	Mono
		STEREO	Korean Stereo	Stereo	Stereo
		MONO + SAP	Sound A Mono	Mono	Mono
		STEREO + SAP	Korean Stereo	Stereo	Stereo
	21	MONO	Sound A Mono	Mono	Mono
		STEREO			
		MONO + SAP	No Matrix	Left = Mono Right = SAP	Left = Mono Right = SAP
		STEREO + SAP			
FM-Radio	40	MONO	Sound A Mono	Mono	Mono
		STEREO	Korean Stereo	Stereo	Stereo

Table 6–12: Manual Sound Select Modes for SAT-reception (FM Matrix is set automatically)

			Source Channels of Sound Select Block for SAT-Modes			
Broadcasted Sound Standard	Selected MSP Standard Code	Broadcasted Sound Mode	FM/AM (source select: 0)	Stereo or A/B (source select: 1)	Stereo or A (source select: 3)	Stereo or B (source select: 4)
FM SAT	6, 50 _{hex}	MONO	Mono	Mono	Mono	Mono
	51 _{hex}	STEREO	Stereo	Stereo	Stereo	Stereo
		BILINGUAL	Left = A (FM1) Right = B (FM2)	Left = A (FM1) Right = B (FM2)	A (FM1)	B (FM2)

7. Appendix C: Application Information

7.1. Exclusions of Audio Baseband Features

In general, all functions can be switched independently. Two exceptions exist:

1. NICAM cannot be processed simultaneously with secondary channel (see Fig. 2–3 and Fig. 2–2 on page 10).
2. FM adaptive deemphasis cannot be processed simultaneously with FM-identification.

7.2. Phase Relationship of Analog Outputs

The analog output signals: Main, Aux, and SCART2 all have the same phases. The SCART1 output has opposite phase.

Using the I²S-outputs for other DSPs or D/A converters, care must be taken to adjust for the correct phase.

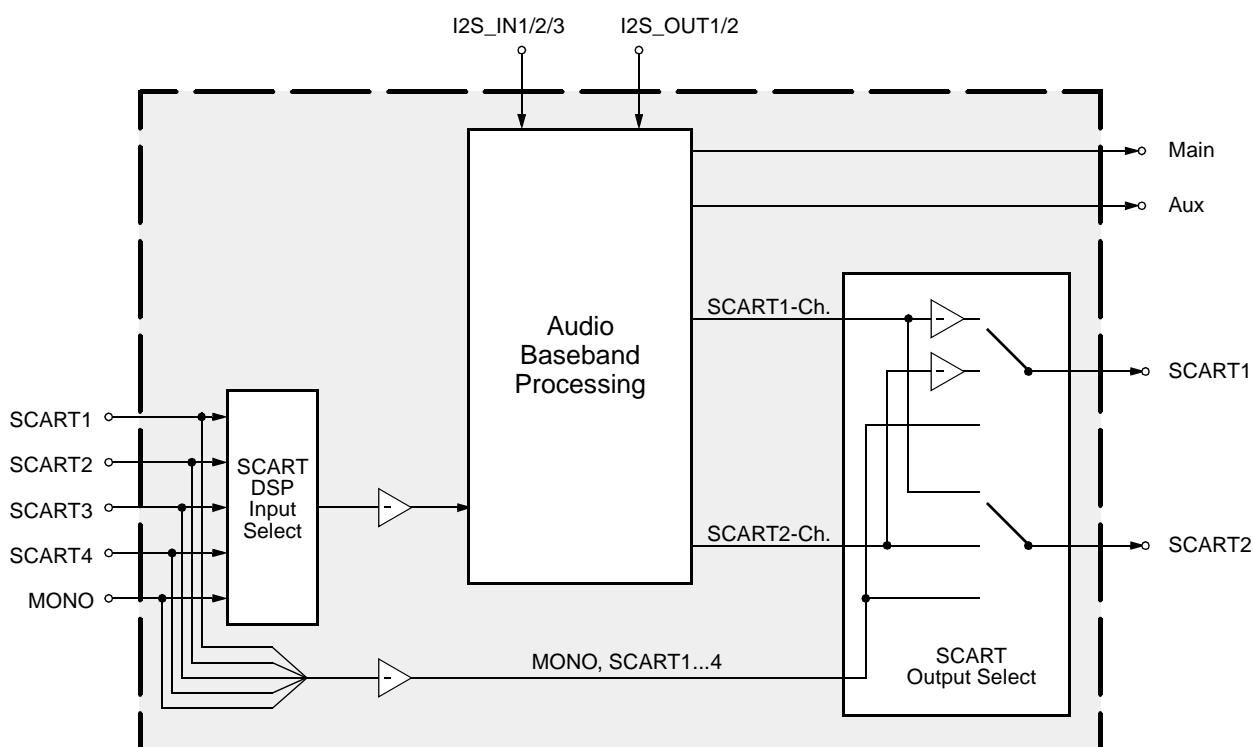
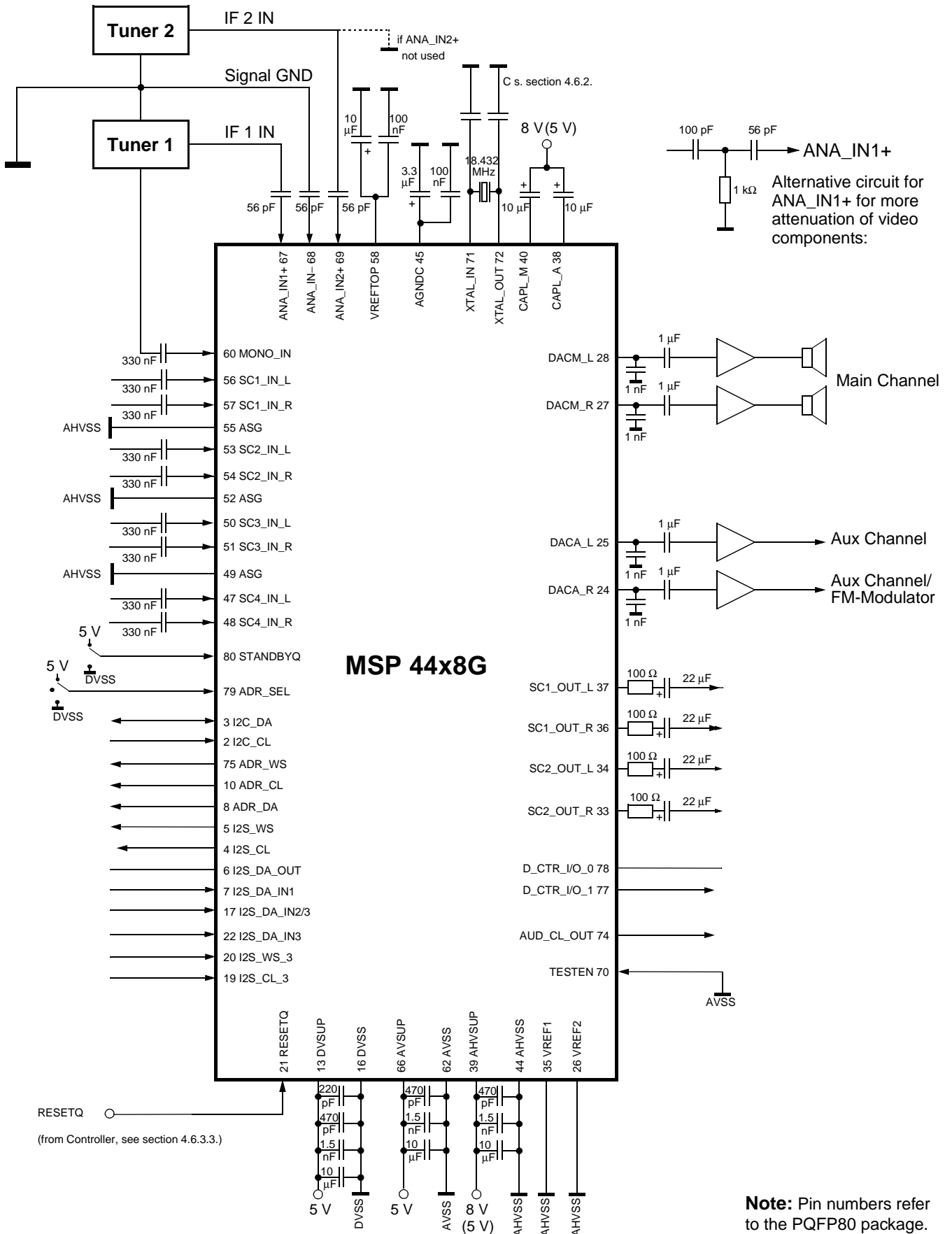


Fig. 7–1: Phase diagram of the MSP 44x8G

7.3. Application Circuit



8. Data Sheet History

1. Preliminary data sheet: "MSP 44x8G Multistandard Sound Processor Family, Feb. 25, 2000, 6251-516-1PD. First release of the preliminary data sheet.

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Printed in Germany
Order No. 6251-516-1PD

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