Preliminary Data Sheet Supplement

Subject:	Version Change
Data Sheet Concerned:	MSP 34x2G 6251-520-1PD, Edition May 22, 2000
Supplement:	No. 1/ 6251-520-1PDS
Edition:	Oct. 18, 2000

Changes from the MSP 34x2G Version A1 to the MSP 34x2G Version A2:

1. The MSP 34x2G-A2 is hardware and software compatible to the MSP 34x2G-A1.

Attachment: Additional Information for MSP 34x2G-A2

MSP 34x2G-A2
Multistandard Sound Processor
**with Dolby Surround Pro Logic**

1. Introduction

This document describes additional features of the MSP 34x2G-A2 in respect to the MSP 34x2G-A1.

1.1. Features

– Improved Automatic Volume Control (AVC) with adjustable reference level, maximum gain, and maximum attenuation

– Additional Multichannel Center mode for TVs with external L/R loudspeakers

– Signal feedback paths for main and aux signals

– Three-channel five-band equalizer in multichannel mode

– Freely programmable Acoustical Compensation Filters (ACF) for loudspeaker and center signals

– Additional loudspeaker mute for TVs with external loudspeakers

– Channel inversion for loudspeaker outputs

2. Architecture
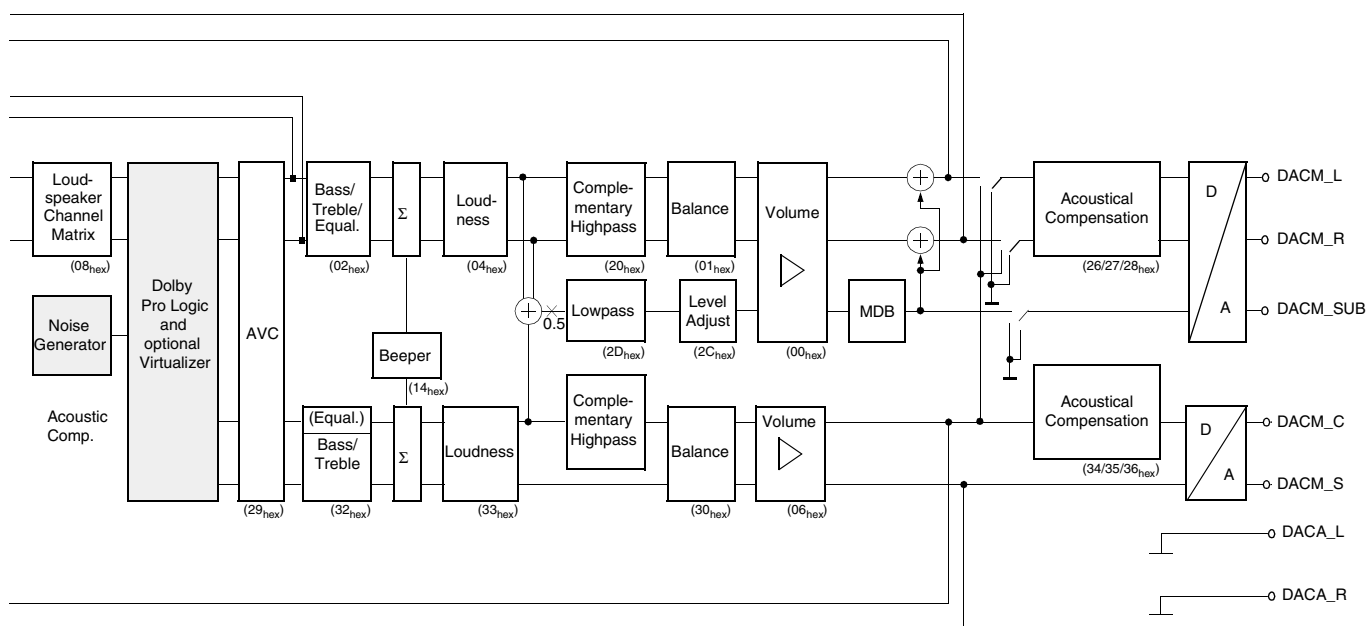
Fig. 2–1: Output section with multi-channel surround

3. Control Interface

Note: Numbers listed in column "See Page" refer to the MSP 34x2G data sheet (6251-520-1PD).

Table 3–1: List of MSP 34x2G Write Registers

Write Register	Address (hex)	Bits	Description and Adjustable Range	Reset	See Page
I^2C Subaddress = 10_{hex} ; Registers are not readable					
STANDARD SELECT	00 20	[15:0]	Initial Programming of complete Demodulator	00 00	28
MODUS	00 30	[15:0]	Demodulator, Automatic and I^2S options	00 00	30
I2S CONFIGURATION	00 40	[15:0]	Configuration of I^2S format	00 00	31
I^2C Subaddress = 12_{hex} ; Registers are all readable by using I^2C Subaddress = 13_{hex}					
Volume loudspeaker channel	00 00	[15:8]	[+12 dB ... −114 dB, MUTE]	MUTE	35
Volume / Mode loudspeaker channel		[7:0]	1/8 dB Steps, Reduce Volume / Tone Control / Compromise	00_{hex}	
Balance loudspeaker channel [L/R]	00 01	[15:8]	[0...100 / 100% and 100 / 0...100%] [−127...0 / 0 and 0 / −127...0 dB]	100% / 100%	36
Balance mode loudspeaker		[7:0]	[Linear mode / logarithmic mode]	linear mode	
Bass loudspeaker channel	00 02	[15:8]	[+20 dB ... −12 dB]	0 dB	37
Treble loudspeaker channel	00 03	[15:8]	[+15 dB ... −12 dB]	0 dB	38
Loudness loudspeaker channel	00 04	[15:8]	[0 dB ... +17 dB]	0 dB	39
Loudness filter characteristic		[7:0]	[NORMAL, SUPER_BASS]	NORMAL	
Spatial effect strength loudspeaker ch.	00 05	[15:8]	[−100%...OFF...+100%]	OFF	40
Spatial effect mode/customize		[7:0]	[SBE, SBE+PSE]	SBE+PSE	
Volume headphone *) channel	00 06	[15:8]	[+12 dB ... −114 dB, MUTE]	MUTE	35
Volume / Mode headphone *) channel		[7:0]	1/8 dB Steps, Reduce Volume / Tone Control	00_{hex}	
Volume SCART1 output channel	00 07	[15:8]	[+12 dB ... −114 dB, MUTE]	MUTE	41
Loudspeaker source select	00 08	[15:8]	[FM/AM, NICAM, SCART, I^2S1, I^2S2]	FM/AM	6
Loudspeaker channel matrix		[7:0]	[SOUNDA, SOUNDB, STEREO, MONO...]	SOUNDA	34
Headphone *) source select	00 09	[15:8]	[FM/AM, NICAM, SCART, I^2S1, I^2S2]	FM/AM	6
Headphone *) channel matrix		[7:0]	[SOUNDA, SOUNDB, STEREO, MONO...]	SOUNDA	34
SCART1 source select	00 0A	[15:8]	[FM/AM, NICAM, SCART, I^2S1, I^2S2]	FM/AM	6
SCART1 channel matrix		[7:0]	[SOUNDA, SOUNDB, STEREO, MONO...]	SOUNDA	34
I^2S source select	00 0B	[15:8]	[FM/AM, NICAM, SCART, I^2S1, I^2S2]	FM/AM	6
I^2S channel matrix		[7:0]	[SOUNDA, SOUNDB, STEREO, MONO...]	SOUNDA	34
Quasi-peak detector source select	00 0C	[15:8]	[FM/AM, NICAM, SCART, I^2S1, I^2S2]	FM/AM	6
Quasi-peak detector matrix		[7:0]	[SOUNDA, SOUNDB, STEREO, MONO...]	SOUNDA	34
Prescale SCART input	00 0D	[15:8]	[00_{hex} ... $7F_{hex}$]	00_{hex}	34
Prescale FM/AM	00 0E	[15:8]	[00_{hex} ... $7F_{hex}$]	00_{hex}	32
FM matrix		[7:0]	[NO_MAT, GSTERERO, KSTEREO]	NO_MAT	33
Prescale NICAM	00 10	[15:8]	[00_{hex} ... $7F_{hex}$] (MSP 3412G, MSP 3452G only)	00_{hex}	33
Prescale I^2S2	00 12	[15:8]	[00_{hex} ... $7F_{hex}$]	10_{hex}	33

Table 3–1: List of MSP 34x2G Write Registers, continued

Write Register	Address (hex)	Bits	Description and Adjustable Range	Reset	See Page
ACB: SCART Switches a. D_CTR_I/O	00 13	[15:0]	Bits[15:0]	00_{hex}	42
Beeper	00 14	[15:0]	$[00_{hex} ... 7F_{hex}]/[00_{hex} ... 7F_{hex}]$	0/0	42
Prescale I²S1	00 16	[15:8]	$[00_{hex} ... 7F_{hex}]$	10_{hex}	33
Mode tone control	00 20	[15:8]	[BASS/TREBLE, EQUALIZER]	BASS/TREB	37
Equalizer loudspeaker ch. band 1	00 21	[15:8]	[+12 dB ... −12 dB]	0 dB	38
Equalizer loudspeaker ch. band 2	00 22	[15:8]	[+12 dB ... −12 dB]	0 dB	38
Equalizer loudspeaker ch. band 3	00 23	[15:8]	[+12 dB ... −12 dB]	0 dB	38
Equalizer loudspeaker ch. band 4	00 24	[15:8]	[+12 dB ... −12 dB]	0 dB	38
Equalizer loudspeaker ch. band 5	00 25	[15:8]	[+12 dB ... −12 dB]	0 dB	38
Acoustical Compensation loudspeaker	00 26	[15:0]	C0_Main	0	
Acoustical Compensation loudspeaker	00 27	[15:0]	C1_Main	0	
Acoustical Compensation loudspeaker	00 28	[15:0]	C2_Main	0	
Automatic Volume Correction	00 29	[15:8]	[off, on, decay time]	off	6
		[7:0]	[output level; max gain; max attenuation]	[−18,6,24]	6
loudspeaker channel mute and invert	00 2B	[7:0]	[on, invert, mute]	on	41
Subwoofer level adjust	00 2C	[15:8]	[0 dB ... −30 dB, mute]	0 dB	41
Subwoofer corner frequency	00 2D	[15:8]	[50 Hz ... 400 Hz]	00_{hex}	41
Subwoofer complementary high-pass		[7:0]	[off, on]	off	41
Balance headphone*) channel [L/R]	00 30	[15:8]	[0...100 / 100% and 100 / 0...100%] [−127...0 / 0 and 0 / −127...0 dB]	100 %/100 %	36
Balance mode headphone*)		[7:0]	[Linear mode / logarithmic mode]	linear mode	
Bass headphone*) channel	00 31	[15:8]	[+20 dB ... −12 dB]	0 dB	37
Treble headphone*) channel	00 32	[15:8]	[+15 dB ... −12 dB]	0 dB	38
Loudness headphone*) channel	00 33	[15:8]	[0 dB ... +17 dB]	0 dB	39
Loudness filter characteristic *)		[7:0]	[NORMAL, SUPER_BASS]	NORMAL	
Acoustical Compensation center	00 34	[15:0]	C0_Center	0	
Acoustical Compensation center	00 35	[15:0]	C1_Center	0	
Acoustical Compensation center	00 36	[15:0]	C2_Center	0	
Volume SCART2 output channel	00 40	[15:8]	[+12 dB ... −114 dB, MUTE]	00_{hex}	41
SCART2 source select	00 41	[15:8]	[FM, NICAM, SCART, I²S1, I²S2]	FM	6
SCART2 channel matrix		[7:0]	[SOUNDA, SOUNDB, STEREO, MONO...]	SOUNDA	34
AUX/CS switch	00 48	[15]	[AUX, CS]	0_{hex}	43
Channel configuration		[14:8]	[STEREO/TWO_CHANNEL/MULTI_CHANNEL/ MULTI_CHANNEL_CENTER]	00_{hex}	43
Mode tone control center channel		[7:0]	[BASS/TREBLE, EQUALIZER]	BASS/TREB	43
Spatial effect for surround processing	00 49	[15:8]	[0% - 100%]	00_{hex}	43
Virtual surround effect strength	00 4A	[15:8]	[0% - 100%]	00_{hex}	44

Table 3–1: List of MSP 34x2G Write Registers, continued

Write Register	Address (hex)	Bits	Description and Adjustable Range	Reset	See Page
Decoder matrix	00 4B	[15:8]	[ADAPTIVE/PASSIVE/EFFECT]	00_{hex}	44
Surround reproduction		[7:4]	[REAR_SPEAKER/FRONT_SPEAKER/PANORAMA/ 3D_PANORAMA]	0_{hex}	44
Center mode		[3:0]	[PHANTOM/NORMAL/WIDE/OFF]	0_{hex}	44
Surround delay	00 4C	[15:0]	[5..31 ms]	00_{hex}	45
Noise Generator	00 4D	[15:0]	[NOISEL, NOISEC, NOISER, NOISES]	00_{hex}	45
[*) In Multi-Channel Mode, these registers are used for controlling baseband functions of the center and surround channels. The following relationship applies: Center corresponds to the left headphone channel, Surround corresponds to the right headphone channel.]					

Table 3–2: Write Registers on I^{2}C Subaddress 12$_{hex}$

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$_{hex}$	**Source for:** 　　　　**Loudspeaker Output** 　　　　**Headphone Output** 　　　　**SCART1 DA Output** 　　　　**SCART2 DA Output** 　　　　**I^{2}S Output** 　　　　**Quasi-Peak Detector**			SRC_MAIN SRC_AUX SRC_SCART1 SRC_SCART2 SRC_I2S SRC_QPEAK
	bit[15:8]	0	"FM/AM": demodulated FM or AM mono signal	
		1	"Stereo or A/B": demodulator Stereo or A/B signal (in manual mode, this source is identical to the NICAM source in the MSP 3410D)	
		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	I^{2}S1 input	
		6	I^{2}S2 input	
		12	Main channel: AVC processed signal	
		13	Main channel: baseband processed signal with volume	
		14	Aux channel: baseband processed signal with volume	
00 29$_{hex}$	**Automatic Volume Correction (AVC) Loudspeaker Channel**			
	bit[15:12]	00$_{hex}$ 08$_{hex}$	AVC off (and reset internal variables) AVC on	AVC
	bit[11:8]	decay time 08$_{hex}$ 04$_{hex}$ 02$_{hex}$ 01$_{hex}$	 8 s decay time 4 s decay time 2 s decay time 20 ms decay time (should be used for approx. 100 ms after channel change)	AVC_DECAY
	bit[7:4]	output level 0$_{hex}$ 1$_{hex}$ ... f$_{hex}$	 −18 dBFS −17 dBFS −3 dBFS	AVC_LEVEL
	bit[3:2]	maximum attenuation 0$_{hex}$ 1$_{hex}$ 2$_{hex}$	 24 dB 18 dB 12 dB	AVC_MIN
	bit[1:0]	maximum gain 0$_{hex}$ 1$_{hex}$ 3$_{hex}$	 6 dB 12 dB 0 dB	AVC_MAX

Table 3–2: Write Registers on I^{2}C Subaddress 12$_{hex}$, continued

Register Address	Function	Name
00 2B$_{hex}$	**Mute or Invert Loudspeaker D/A Output** bit[15:2] must be zero bit[1:0] $\quad$ 0$_{hex}$ $\qquad$ no modification $\qquad\qquad$ 1$_{hex}$ $\qquad$ invert left channel of D/A output $\qquad\qquad$ 2$_{hex}$ $\qquad$ mute D/A ouput	MUT_INV_M
00 26$_{hex}$ 00 27$_{hex}$ 00 28$_{hex}$ 00 34$_{hex}$ 00 35$_{hex}$ 00 36$_{hex}$	**Acoustical Compensation Filter Loudspeaker Channel:** $\qquad$ **C0_Main** $\qquad$ **C1_Main** $\qquad$ **C2_Main** **Acoustical Compensation Filter Headphone Left Channel (Center):** $\qquad$ **C0_Center** $\qquad$ **C1_Center** $\qquad$ **C2_Center** These cells determine the coefficients of a second order filter for acoustical compensation of loudspeaker responses. The transfer function of this filter is $$H(z) \;=\; \frac{(1 - a0 + 2 \times a1 \times z^{-1} + a2 \times z^{-2})}{(1 + 2 \times b1 \times z^{-1} + b2 \times z^{-2})}$$ The transfer function must not have more than 0 dB gain. Micronas will supply a design tool for these coefficients. This feature is switched off by setting all coefficients to zero (reset state). A mute or fastmute operation should precede any change of these coefficients. The coefficients are two's complement numbers ranging from $[-1.0...1.0]$. **C0:** bit[15:6] $\quad$ 10-bit coefficient a0 bit[5:3] $\quad$ 3 LSBs for coefficient b1 (together with 6 bit of c1, this forms a 9-bit coefficient for b1) bit[2:0] $\quad$ 3 LSBs for coefficient b2 (together with 6 bit of c2, this forms a 9-bit coefficient for b2) **C1:** bit[15:6] $\quad$ 10-bit coefficient a1 bit[5:0] $\quad$ 6 MSBs for coefficient b1 **C2:** bit[15:6] $\quad$ 10-bit coefficient a2 bit[5:0] $\quad$ 6 MSBs for coefficient b2	ACF_M0 ACF_M1 ACF_M2 ACF_C0 ACF_C1 ACF_C2

Table 3–2: Write Registers on I^{2}C Subaddress 12$_{hex}$, continued

Register Address	Function	Name
00 48$_{hex}$	**Main/Aux Configuration**	MA_CONF

	bit[15] AUX/CS Switch
	            0 Aux (Headphone) outputs are active (pin names: DACA_L, DACA_R), CS outputs are muted
	            1 CS (Center/Surround) outputs are active (pin names: DACM_C, DACM_S), Aux outputs are muted (C corresponds to Aux L, S to Aux R)
		AUX_CS

	The AUX/CS switch defines which output pin pair is driven by the D/A converters that are used for headphone or surround processing. The unselected pins are muted. This makes it convenient to connect the center/surround amplifiers or outputs to the MSP 34x2G without external switches. The Headphone/Surround channel should be muted before switching (set register 06$_{hex}$ to: 0000$_{hex}$). Allow at least 2 s for settling.	

	bit[14:8] Channel Configuration
	        00$_{hex}$ STEREO: This mode is used in plain stereo mode. Standard processing applies to the loudspeaker and headphone channels. Surround processing is switched off. In this mode, the IC is compatible to the MSP 3450G (if bit[15] is equal to 0).
	        01$_{hex}$ TWO_CHANNEL: This mode is used for virtual surround sound. The surround processing block is active and its left and right outputs are distributed to the loudspeaker output channel. The processing on the headphone channel remains standard. In this mode, the IC is comparable to the MSP 3451G.
	        02$_{hex}$ MULTI_CHANNEL: This mode is used for surround sound with more than 2 channels. The surround processing block is active and its left and right outputs are distributed to the loudspeaker output channel, its center and surround outputs are distributed to the headphone output channel. No headphone processing is possible. In this mode, it is convenient to select the C/S pins by setting bit[15] to 1.
	        03$_{hex}$ MULTI_CHANNEL_CENTER: This mode is used for surround sound with more than 2 channels. The surround processing block is active and its left and right outputs are distributed to the loudspeaker output channel, its center and surround outputs are distributed to the headphone output channel. Just after the volume control, the center signal is distributed to the left and right loudspeaker outputs as well as to the center outputs. The left and right signals can be accessed via the feedback path to the source selector. No headphone processing is possible. In this mode, it is convenient to select the C/S pins by setting bit[15] to 1.
		CHAN_CONF

	bit[7:0] Mode Tone control center channel
	        00$_{hex}$ Bass/treble for center channel
	        01$_{hex}$ The center signal is processed with an equalizer. The same band setting as for the loudspeaker equalizer is used. The surround channel is processed with bass/treble. This mode is only allowed in Channel Configurations 2 and 3.	

Table 3–3: Read Registers on I^{2}C Subaddress 13$_{hex}$

Register Address	Function	Name
	MSP 34X2G VERSION READOUT REGISTERS	
00 1E$_{hex}$	**MSP Hardware Version Code** bit[15:8] 01$_{hex}$ MSP 34x2G - <u>A</u>2 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. **MSP Major Revision Code** bit[7:0] 07$_{hex}$ MSP 34x2<u>G</u> - A2 The major revision code of the MSP 34x2G is 7.	MSP_HARD MSP_REVISION

00 1F$_{hex}$	**MSP Product Code**
	
	bit[15:8] 02$_{hex}$ MSP 34<u>02</u>G - A2
	            0C$_{hex}$ MSP 34<u>12</u>G - A2
	            16$_{hex}$ MSP 34<u>22</u>G - A2
	            2A$_{hex}$ MSP 34<u>42</u>G - A2
	            34$_{hex}$ MSP 34<u>52</u>G - A2
	
	By means of the MSP-Product Code, the control processor is able to decide which TV sound standards and audio baseband features have to be considered.
	
	MSP ROM Version Code
	
	bit[7:0] 42$_{hex}$ MSP 34x2G - A<u>2</u>
	
	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 34x2G versions according to this number.
	
	To avoid compatibility problems with MSP 3410B and MSP 34x0D, an offset of 40$_{hex}$ is added to the ROM version code of the chip's imprint.
		MSP_PRODUCT
		
		MSP_ROM