NJU25007

VMAx™ 3D VirtualTheater Digital Audio Processor

VMAx VIRTUAL DOLBY SURROUND DOLBY SURROUND PRO • LOGIC

Description

The NJU25007 is a digital surround sound processor which decodes surround information from a stereo audio source and produces accurate and realistic 3D sonic imaging from two speakers. A full front and rear channel effect is generated using VMAx™ (short for Virtual Multi-Axis sound) 3D Virtual-Sound. VMAx has the ability to make specific sounds appear to come from any point in three dimensional space surrounding the listener. VMAx uses HRTF (Head Related Transfer Function) processing on surround, center and front channels along with precise digital filtering to provide high fidelity audio without added artifacts, coloration, or compromise of the original source. The excellent audio quality of the NJU25007 makes it suitable for use in high-end stereo pre-amps and A/V receivers. At the same time the NJU25007 is cost effective for consumer products such as TV's, multimedia computers, and arcade games.

The NJU25007 is also a Dolby™ Pro Logic™ decoder capable of six outputs: Front Left and Right, Surround Left and Right, Center, and Subwoofer. All this is possible using a 24-bit DSP core developed by Medianix for this application which contains all the necessary ROM and RAM internal to the chip. The DSP program code is contained in the on-chip ROM.

VMAx is implemented by using the five channel output from the Dolby Pro Logic decoder in the NJU25007. The HRTF's (Head Related Transfer Functions) are a series of concatenated filters designed to compensate for speaker response and position. The desired spatial sound information is created by proprietary coefficients which are a function of the angle of separation of the actual speakers and the desired position of the virtual sound image. The parameters are optimized for actual front speakers placed ±20° apart and virtual speakers at 0°, ±40°, and ±90° around the listener position.

Features

- VMAx™ 3D VirtualTheater Audio Surround Processing
  - Highly effective 360° surround sound from 2 speakers
  - Hi fidelity audio faithfully preserved
  - Based on the original Cooper-Bauck 3D technology
- Dolby Pro Logic Decoder With Optional Subwoofer Output
  - 5.0 or 5.1ch output (L,R,SL,SR,C,SW)
  - On-chip digital surround delay, 30ms max.
- Exclusive Medianix MxD™ Decorrelation Technology
  - Expands Pro Logic mono surround into stereo
  - Synthesizes mono-input to stereo-output
- Dolby 3 Stereo Mode With Optional Subwoofer Output
  - 3.0 or 3.1ch output (L,R,C,SW)
- System Level Audio Management
  - Tone control
    - Independent bass and treble
    - +15dB to -16dB in 1dB steps
  - Bass Management
    - Subwoofer filter (80Hz default)
    - Front high pass filters
    - Programmable cutoff frequencies
  - Master Volume Control
  - Clipping Indicator
- A complete DSP-based 3D Audio Solution
  - Includes on-chip surround time delay

Figure 1 VMAx 3D VirtualTheater

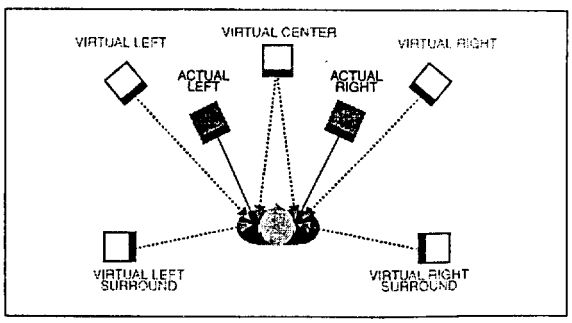


Figure 2 MED25007 Pin Configuration

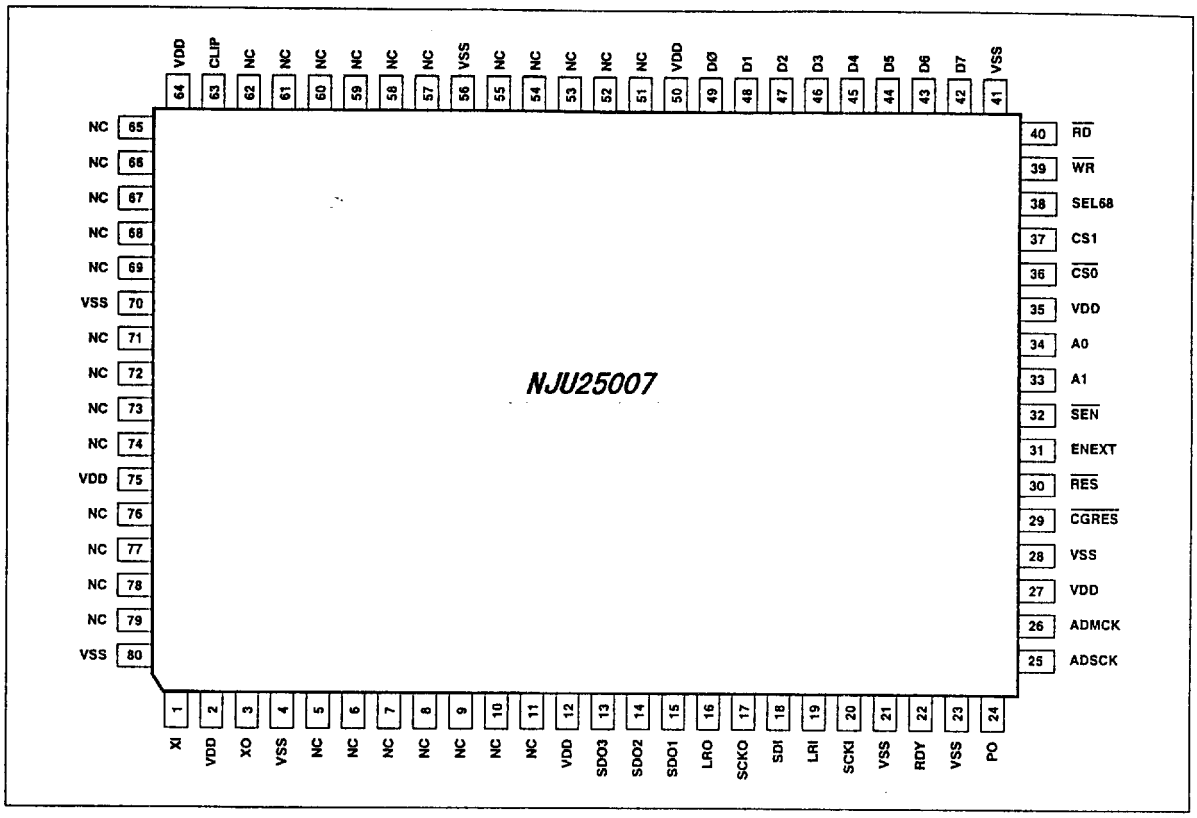


Figure 3 NJU25007 Functional Block Diagram

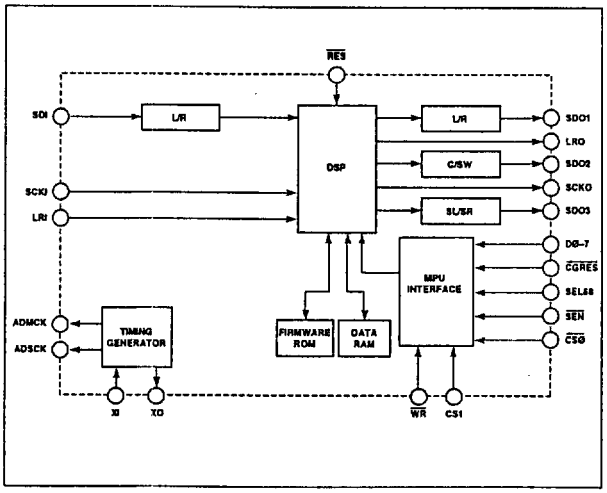
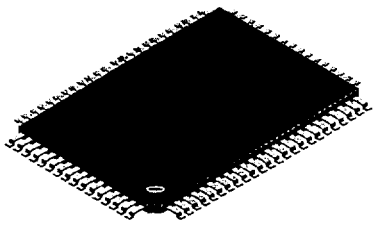


Figure 4 VMAx QFP-80 Package



Plastic QFP-80

Table 1 Pin Description

No.	Symbol	I/O	Function	No.	Symbol	I/O	Function
1	XI	I	Crystal/External clock input	41	VSS	I	Ground
2	VDD	I	Power supply terminal: +5V	42	D7	I	MPU data, parallel input (MSB)
3	XO	O	Crystal	43	D6	I	MPU data, parallel input
4	VSS	I	Ground	44	D5	I	MPU data, parallel input
5	NC		No connect	45	D4	I	MPU data, parallel input
6	NC		No connect	46	D3	I	MPU data, parallel input
7	NC		No connect	47	D2	I	MPU data, parallel input
8	NC		No connect	48	D1		MPU data, parallel input
9	NC		No connect	49	D0	I	MPU data, parallel input (LSB), serial data input ($\overline{SEN}$ = 0)
10	NC		No connect	50	VDD	I	Power supply terminal: +5V
11	NC		No connect	51	NC		No connect
12	VDD	I	Power supply terminal: +5V	52	NC		No connect
13	SDO3	O	Digital audio serial data out, surround right and left	53	NC		No connect
14	SDO2	O	Digital studio serial data out, center and subwoofer	54	NC		No connect
15	SDO1	O	Digital audio serial data out, front right and left	55	NC		No connect
16	LRO	O	Output left/right frame clock	56	VSS	I	Ground
17	SCKO	O	Output digital audio serial clock	57	NC		No connect
18	SDI	I	Input digital audio serial data	58	NC		No connect
19	LRI	I/O	Input left/right frame clock	59	NC		No connect
20	SCKI	I/O	Input digital audio serial clock	60	NC		No connect
21	VSS	I	Ground	61	NC		No connect
22	RDY	I	Test pin, high for normal operation	62	NC		No connect
23	VSS	I	Ground	63	CLIP	O	Clipping Indicator
24	PO	O	Test pin	64	VDD	I	Power supply terminal: +5V
25	ADSCK	O	32Fs/64Fs serial clock for A/D, D/A converters (default 32Fs)	65	NC		No connect
26	ADMCK	O	384Fs/256Fs master clock for A/D, D/A converters (default 384Fs)	66	NC		No connect
27	VDD	I	Power supply terminal: +5V	67	NC		No connect
28	VSS	I	Ground	68	NC		No connect
29	$\overline{CGRES}$	I	Test pin, normally high	69	NC		No connect
30	$\overline{RES}$	I	Reset, must be held low for at least two clock cycles after power on	70	VSS	I	Ground
31	ENEXT	I	Test pin, normally low	71	NC		No connect
32	$\overline{SEN}$	I	MPU serial interface enable. Serial = L, parallel = H	72	NC		No connect
33	A1	I	Test pin. Low for normal operation	73	NC		No connect
34	A0	I	Test pin. Low for normal operation	74	NC		No connect
35	VDD	I	Power supply terminal: +5V	75	VDD	I	Power supply terminal: +5V
36	$\overline{CS0}$	I	Chip select, MPU interface enabled when $\overline{CS0}$ = 0 and CSI = 1	76	NC		No connect
37	CS1	I	Chip select, MPU interface enabled when $\overline{CS0}$ = 0 and CS1 = 1	77	NC		No connect
38	SEL68	I	MPU interface mode: 68K = H, Z80 = L	78	NC		No connect
39	$\overline{WR}$	I	Write Strobe (Z80), Write Enable (68K)	79	NC		No connect
40	$\overline{RD}$	I	Write Strobe (68K)	80	VSS	I	Ground

Specifications

Table 2 Absolute Maximum Ratings

Parameter	Symbol	Minimum	Maximum	Unit
Supply Voltage (T_A = 25°C)	V_{DD}	-0.3	7	V
Input, Output Pin Voltage (T_A= 25°C)	V_x	-0.3	V_{DD} + 0.3	V
Operating Temperature	t_{OPR}	-20	70	°C
Storage Temperature	t_{STG}	-55	125	°C

Table 3 Electrical Characteristics (T_A = -20°C to +70°C)

Parameter	Symbol	Test Condition	Minimum	Typical	Maximum	Unit
Operating Voltage	V_{DD}	V_{DD} pins	4.75		5.25	V
Operating Current	I_{DD}	f_{osc} = 37 MHz		90	125	mA
High Level Input Voltage	V_{IH}		0.80 V_{DD}		V_{DD}	V
Low Level Input Voltage	V_{IL}		V_{SS}		0.10 V_{DD}	V
High Level Input Current	I_{IH}	V_{IN} = V_{DD}			10	µA
Low Level Input Current	I_{IL}	V_{IN} = V_{SS}			10	µA
High Level Output Voltage	V_{OH}	I_{OH} = 2 mA	V_{DD} - 1.0			V
Low Level Output Voltage	V_{OL}	I_{OH} = 2 mA			0.5	V
Input Capacitance	C_{IN}			10	20	pF
Clock Frequency	f_{OSC}		20		37	MHz
Ext. System Clock Duty Cycle	r_{EC}		45		55	%

Table 4 Serial Data Input Timing Parameters (V_{DD} = 5V ± 5%, T_A = -20°C to +70°C, f_{CLK} = 37MHz)

Parameter	Symbol	Test Condition	Minimum	Typical	Maximum	Unit
SCKI Period			160			
L Pulse Width	t_{SIL}		80			ns
H Pulse Width	t_{SIH}		80			
SCKI to LRI Time	t_{SLI}		50			ns
LRI to SCKI Time	t_{LSI}		75			ns
Data Setup Time	t_{DS}		10			ns
Data Hold Time	t_{DH}		10			ns

Table 5 Serial Data Output Timing Parameters (V_{DD} = 5V ± 5%, T_A = -20°C to +70°C, f_{CLK} = 37MHz)

Parameter	Symbol	Test Condition	Minimum	Typical	Maximum	Unit
SCKO Period		C_L: LRO, SCKO, SDO = 5pF	160			ns
L Pulse Width	t_{SOL}		80			
H Pulse Width	t_{SOH}		80			
SCKO to LRO Time	t_{SLO}				5	ns
Data Output Delay	t_{DOD}				5	ns

Table 6 Z80 Interface Timing Parameters (V_{DD} = 5V ± 5%, T_A = -20°C to +70°C, f_{CLK} = 37MHz)

Parameter	Symbol	Test Condition	Minimum	Typical	Maximum	Unit
Address Setup Time	t_{AS8}		100			ns
Address Hold Time	t_{AH8}		100			ns
System Cycle Time	t_{CYC8}		1,000			ns
Read/Write Pulse Width	t_{CC8}		100			ns
Write Data Setup Time	t_{DS8}		10			ns
Write Data Hold Time	t_{DH8}		10			ns

Table 7 68K Interface Timing Parameters (V_{DD} = 5V ± 5%, T_A = -20°C to +70°C, f_{CLK} = 37MHz)

Parameter	Symbol	Test Condition	Minimum	Typical	Maximum	Unit
Address Setup Time	t_{AS6}		100			ns
Address Hold Time	t_{AH6}		100			ns
System Cycle Time	t_{CYC6}		1,000			ns
Read/Write Pulse Width	t_{CC6}		100			ns
Write Data Setup Time	t_{DS6}		10			ns
Write Data Hold Time	t_{DH6}		10			ns
Write-to-read Strobe Setup	t_{WRS}		100			ns
Read-to-write Strobe Setup	t_{WRH}		100			ns

Timing Diagrams

Figure 5 Serial Data Input Timing

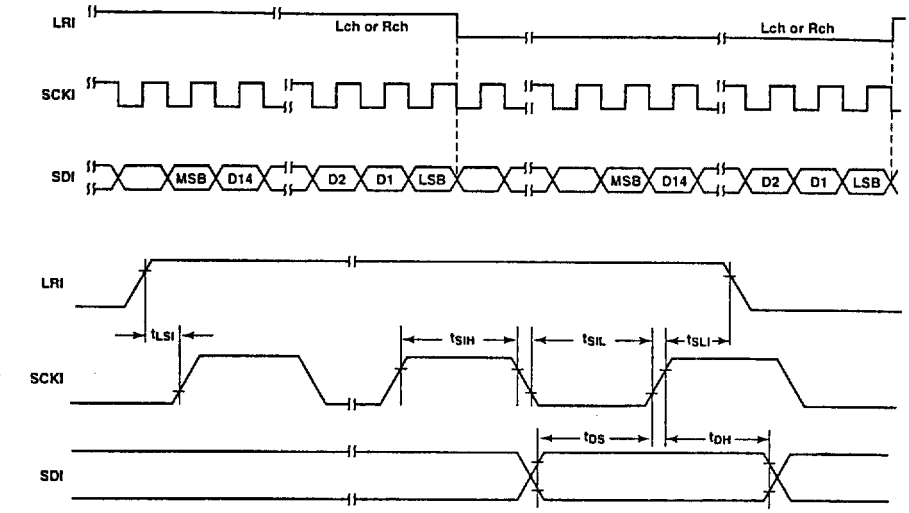


Figure 6 Serial Data Output Timing

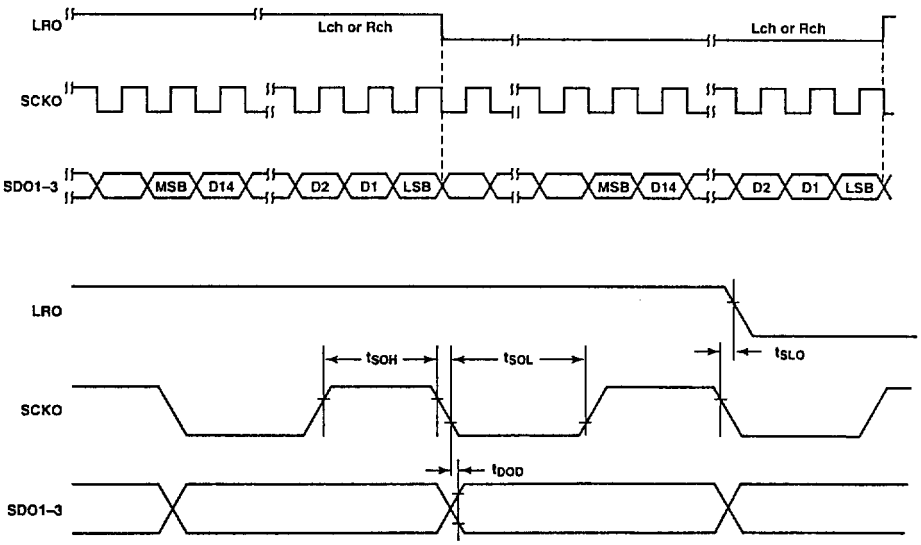


Figure 7 Left-Justified Data Format, ADSCK = 64 Fs, 18-bit Data

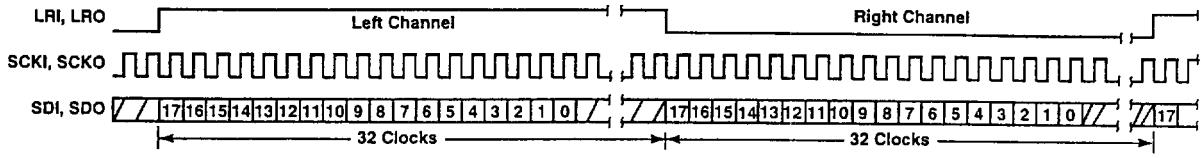


Figure 8 Right-Justified Data Format, ADSCK = 64 Fs, 18-bit Data

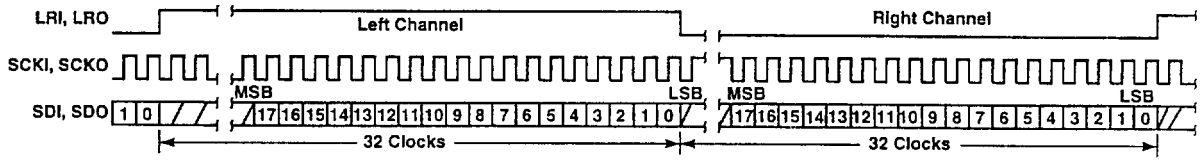


Figure 9 I²S Data Format, ADSCK = 64 Fs, 18-bit Data

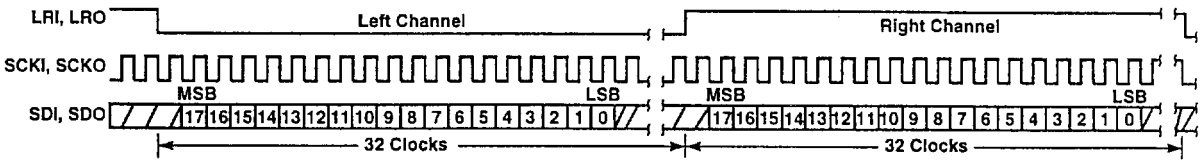


Figure 10 Right- and Left-Justified Data Formats, ADSCK = 32 Fs, 16-bit Data

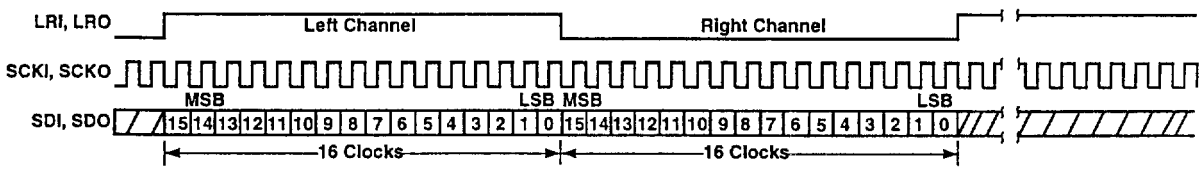


Figure 11 I²S Data Formats, ADSCK = 32 Fs, 16-bit Data

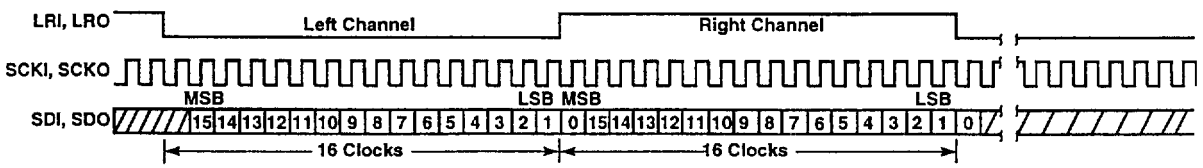


Figure 12 Z80 Parallel Interface Timing (SEL68 Low)

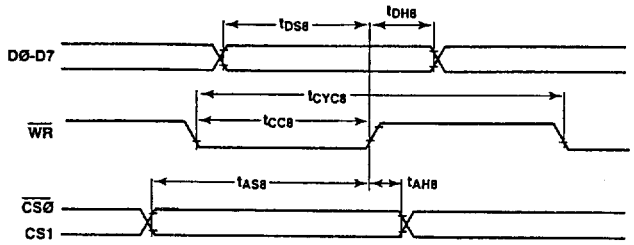


Figure 13 Z80 Serial Interface Timing (SEL68 Low)

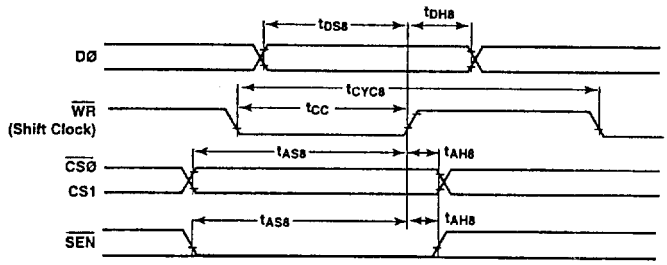


Figure 14 68K Parallel Interface Timing (SEL68 High)

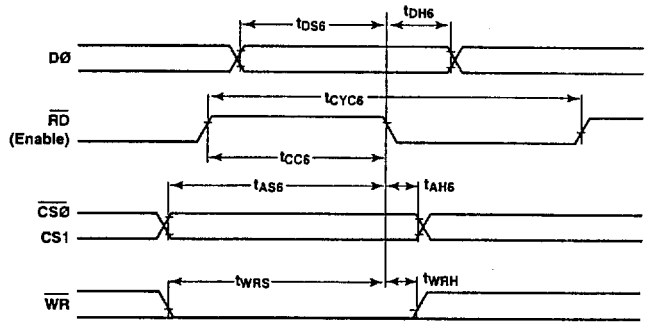
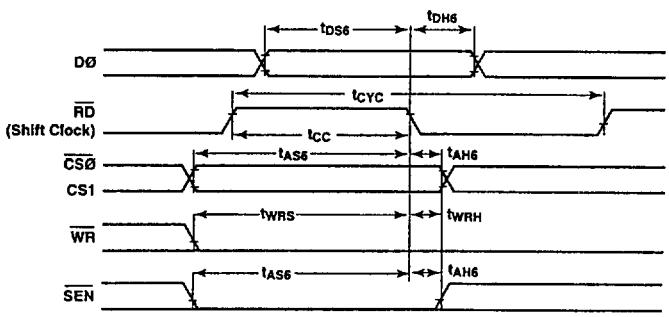


Figure 15 68K Serial Interface Timing (SEL68 High)



Functional Description

High quality Medianix 24-bit DSP technology, and VMAx 3D VirtualTheater and Dolby Pro Logic algorithms in the **NJU25007** together create the effect that rear speaker surround sound audio is present while only two front speakers are active. A digital low pass filtered "0.1 channel" output for a subwoofer is also included. In addition to 2-channel VMAx, the **NJU25007** also supports 4-channel Dolby Pro Logic decoding. In addition, exclusive Medianix MxD™ surround decorrelation technology provides a realistic stereo surround sound signal which can be used while in either mode.

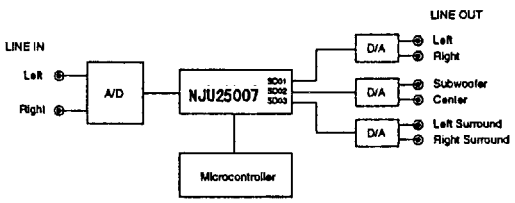
Dolby Pro Logic Surround

Prior to VMAx processing, Dolby Pro Logic decoding of the two channel input signal is required. Pro Logic processes stereo inputs and produces Left, Center, Right, Surround (monophonic) channels. Optionally, a Subwoofer (SW) output is provided which is a low-pass filtered sum of the front output channels.

The **NJU25007** can easily be configured to operate in Pro Logic mode, in which case the Left, Center, Right and Surround outputs as well as the optional Subwoofer (0.1 channel) are available.

The **NJU25007** can be switched under microprocessor control between 2-ch and 6-ch modes. For 4-channel applications using only one surround output feeding both surround speakers and no subwoofer, the Left Surround and Subwoofer data can be swapped in the Configuration Command. This requires only two D/As, or one codec and one D/A on the output for a 4-channel (4.1 ch) surround system with Left, Right, Center, and Surround.

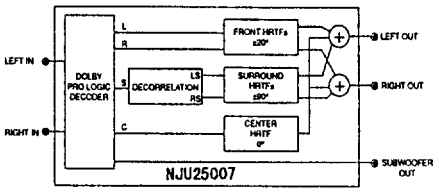
Figure 16 Typical Dolby Pro Logic System



VMAx 3D VirtualTheater Surround Sound

The processing of VMAx 3D surround begins by decoding surround sound encoded audio (preferably matrix-encoded stereo). The on-chip Dolby Pro Logic algorithm results in 4 channel (Left, Center, Right, Surround) surround sound. The mono surround signal is processed by Medianix's MxD surround decorrelation algorithm into a synthesized stereo signal. The VMAx virtualizer algorithm then reprocesses L, C, R, SL, SR channels into a 3D left and right signals. Highly directional encoded source material such as a movie soundtrack results in the most dramatic virtual 3-D sound effect during playback, however all stereo audio sources benefit from VMAx 3D VirtualTheater processing.

Figure 17 VMAx 3D VirtualTheater Mode



Subwoofer Output

The **NJU25007** is equipped to provide an 80Hz low pass filtered subwoofer output, which is derived from the Left, Center and Right channels of the Pro Logic decoder. The cutoff frequency of this optional "0.1 channel" can easily be reconfigured using the Coefficient Download Command.

Clock Generator

A 768Fs crystal wired to pins XI and XO of the **NJU25007** will generate signals that operate the chip's internal circuits as well as signals for external circuits such as A/Ds and D/As. Alternatively, an external clock of 768Fs can be applied to XI, with XO remaining unconnected. Fs can be 32kHz, 44.1 kHz, or 48kHz. The signal at the ADMCK pin is set to either 384Fs or 256Fs, and the signal at the ADSCK pin is set to either 32Fs or 64Fs. The default settings are 384Fs and 32Fs, respectively.

Sample Rate	768Fs DSP Clock
32 kHz	24.576 MHz
44.1 kHz	33.868 MHz
48 kHz	36.864 MHz

Clipping Indicator

Each audio sample is examined for a full-scale or an over-range condition at both the stereo Input and at the Left and Right outputs. Any time the audio data reaches full scale a Clipping Indicator flag is set in the DSP and is

output on pin 24 as an active High logic level, which can be used to turn on a transistor connected to an LED. The Clipping Indicator remains High for approximately 4,000 samples, or about 0.1s after clipping is first detected in order to be visible. When this period has been exceeded without any clipping, the Indicator pin is returned to logic Low. The Clipping Indicator may also be monitored by a microprocessor which can take other appropriate action.

Digital Audio Data Interface

Three digital audio data formats are supported: left justified, right justified, and I²S. The data is always MSB first, 2's complement. Either 16-bit or 18-bit data can be accommodated. The polarities of the L/R clocks (LRI, LRO) are programmable, along with the active edge of the serial bit clocks (SCKI, SCKO). Master clock (ADMCK) and serial bit clock (ADSCK) outputs for the A/D and D/A converters are provided by an internal, programmable clock generator for synchronous operation with the DSP clock (768Fs). Asynchronous data rates are possible as long as the output is slaved to the input and it is close to the three supported sampling frequencies (32kHz, 44.1kHz, and 48kHz).

There is one stereo digital audio input and three stereo digital audio outputs for the L and R main channels, L and R surround channels, Center, and Subwoofer. All three serial data outputs must have identical data formats. In each data format mode, SCLK and LRCLK polarities are independently programmable for input and output. Audio data width (16/18 bits), SCK and MCK frequencies (32/64Fs, 256/384Fs, respectively) must be the same for both input and output.

Serial Data Formats

There are three serial data formats supported for interfacing an A/D and three D/As to the digital audio interface. Either Left Justified, Right Justified, or I²S mode is selected by the FMT0 and FMT1 bits in the second byte of the four-byte System State Download Command. In Left Justified Mode, the MSB is aligned to the edge of LRI or LRO. The data is positioned at the left or "front" side of the L/R pulse (see Figure 7). In Right Justified Mode, the LSB is aligned to the LRI or LRO edge. The data is at the right or "rear" of the L/R pulse (see Figure 8). Sometimes this mode is called Japanese Mode or EIAJ Mode. The I²S Mode is similar to Left Justified Mode, except that the data is delayed one SCLK period and the sense of LRI and LRO is inverted (see Figure 9). In I²S Mode, LRI and LRO must be low for left channel data and high for right channel data, the opposite of other modes. The polarity of LRI and LRO can be inverted independently in any mode by the use of the LRI and LRO.

Normally, the serial data bits generated by a source change on the falling edge of the serial clock so that they can be easily clocked into a shift register on the rising clock edge. Considering an A/D as the source and the **NJU25007** as a destination, the default setting is to clock the serial data input, SDI, in on the rising edge of SCKI. This can be changed to clock data in on the falling edge using the SCKI bit in the System Download Command.

For the output serial data, the **NJU25007** is considered the source and the D/As are considered the destination. The default interface setting is for data to be clocked out of SDO on the falling edge of SCKO so that the D/A clocks data in on the rising edge. This can be changed independently of the input and interface mode using the SCKO bit in the System Download Command.

The MS (Master/Slave) bit in the fourth System State Download Command byte selects either Master Mode or Slave Mode. In Right Justified Mode, Master Mode (MS = 0) is defined such that SCKO and LRO are generated from an internal divider derived from the 768Fs DSP clock (XI). In Slave Mode (MS = 1) SCKO and LRO are the same as SCKI and LRI on the input. This mode should be used for asynchronous data rates, such as data from an S/PDIF receiver connected to the Digital Output from a laser disc player. The output D/A's must be synchronized to the input data by using the low jitter, recovered clock from the S/PDIF receiver to the D/A master clock. When an A/D is used, its master clock should be synchronous to the **NJU25007** using ADMCK at 256Fs or 384Fs. In this case either Master Mode or Slave Mode will work. The default setting is Master Mode (MS = 0).

In I²S Mode, Master and Slave modes have slightly different meanings. Relative to the **NJU25007** Slave Mode (MS = 1) is defined as LRI and SCKI being inputs from the A/D. This means that the A/D must be in Master Mode with L/R and SCLK outputs. Conversely, when the **NJU25007** is in Master Mode (MS = 0), LRI and SCKI are outputs that drive the L/R and SCLK inputs of an A/D operating in slave mode. SCKO and LRO to the D/As are always outputs. The D/A converters can run only in Slave Mode, receiving both the L/R and SCLK from the **NJU25007**. LRO and SCKO are derived from the 768Fs DSP clock. When slave mode is selected, LRO and SCKO are generated by the LRI and SCKI inputs. Slave mode should be used for asynchronous audio data.

When using a codec in slave mode and operating the **NJU25007** in I²S Master Mode, the LRO signal must be used to drive the L/R input of the codec. Although LRI is not used, it must be programmed with the opposite polarity. This is the default setting upon initialization after reset (LRI=0, LRO=1) for typical 2-channel Virtual Dolby Surround applications. For more details, refer to the EVB25006-5 Demonstration Board User's Guide. In all other modes or when using separate A/D and D/A converters that use both LRI and LRO, respectively, the polarity should be the same for LRI and LRO.

For applications that switch between Virtual and Pro Logic Surround modes using a codec for L/R input and front L/R output, and two D/A converters for the other four channels, the L/R input clock for all the converters must operate from the LRO signal generated by the **NJU25007**. The LRI signal is not used when a codec is included, but it must be programmed with opposite polarity. The same applies for a single 6-channel codec. When using a codec and two D/A's, it may be better to use the codec for the surround channels, since the discrete D/A's often have better audio performance, critical for the main Left and Right output. The Left and Right Surround outputs can afford to have lower S/N and THD specifications, characteristic of many low cost codecs.

In Left Justified Mode, only Slave Mode (MS = 0) is allowed. The A/D is required to be in Master Mode and supplies LRI and SCKI to the **NJU25007**. LRO and SCKO are generated from LRI and SCKI inputs.

For each data format, the serial clock frequency for SCKI and SCKO is selected using ADSCK in the System Download Command: either 32Fs (32-bit clocks per sample) or 64Fs (64-bit clocks per sample). SCKI and SCKO must be the same frequency. This clock is generated internally for the SCKI and SCKO in Master Mode, but is also available as an output on the ADSCK pin. It is derived from the 768Fs Input Clock to the **NJU25007**.

If SCKI and SCKO are selected to be 32Fs, the data length must be set to 16-bits (BL = 0) because a stereo pair of 16-bit channels needs all 32 clocks per sample. In this case, both Left and Right Justified Modes look exactly the same (see Figure 10), and either mode setting will work. In I²S mode, the data bits appear shifted by one SCLK and the L/R is inverted (see Figure 11).

An output clock, ADMCK, is derived from the **NJU25007** 768Fs Input Clock for use as an ND or DIA master clock. ADMCK is set to 256Fs or 384Fs using the ADMCK bit in the System Download Command.

Microcontroller Interface

Either a Z80 or 68K type microcontroller can be used to download commands to the **NJU25007**. SEL68 must be set high for 68K and low for Z80. Either serial or 8-bit parallel interface modes may be used, depending on SEN input (L = serial and H = parallel). The MPU interface is enabled when CS0 = 0 and CS = 1, allowing a control signal of either polarity to be used.

Figure 18 Z80 Interface

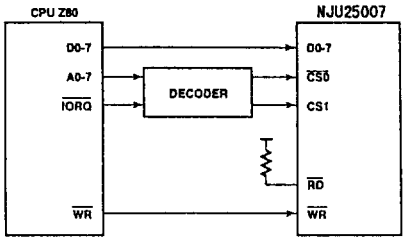
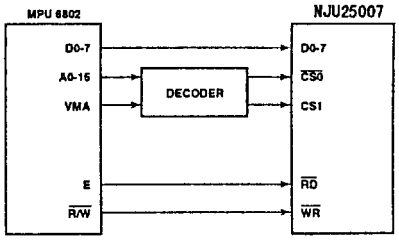


Figure 19 68K Interface



The audio stops when the **NJU25007** receives a System State Download Command or when the Noise Sequencer is turned on. All other commands take immediate effect without interruption in the audio. Only one command should be sent from the microprocessor per L/R period. The DSP in the **NJU25007** spends most of the time processing audio, during which time microprocessor interrupts are disabled. Writing to the MPU interface sets an interrupt request flag inside the chip and holds the data in a register, which is read as soon as the interrupt can be serviced. Successive writes to the MPU interface while the interrupts are disabled will overwrite the previous contents. There is a window of time in each audio sample when it is possible to service several MPU commands. However; there is no way for the microprocessor to determine how long this window is before the audio interrupt arrives and successive MPU writes will be lost. There-

fore, when audio is being processed, it is recommended that only one command be sent to the **NJU25007** for each audio sample, even though the logic will accept parallel writes up to 1 Mbyte per second. When the audio is interrupted (for example, during System State Download Commands), data can be transferred at the maximum data rate, with at least 1 ms between bytes written to the **NJU25007**.

The MPU interface was designed for both read and write operations. However, the DSP firmware only supports write operations.

Parallel Interface

Z80 Mode

Writing commands to the **NJU25007** from a Z80 type microprocessor is accomplished by setting SEL68 low placing data on the input data port, D0 to D7, setting the chip selects, and strobing $\overline{WR}$ low then high. Successive writes, as in the four-byte System Download Command, can be done without resetting the chip select(s). It is recommended to return the Chip Select to the inactive state as soon as the command is sent. The recommended sequence for writing parallel data to the **NJU25007** is:

1. Set $\overline{RD}$ high (can be tied to Vcc).
2. Place data on the data port, D0:7.
3. Assert chip selects, ($\overline{CS0}$ = 0 and CS1 = 1).
   One can be permanently tied to its active state, while the other is controlled.
4. Strobe $\overline{WR}$ low, then high.
5. Deactivate chip selects.
6. Wait at least eight audio samples before sending another command.

68K Mode

In 68K mode (SEL68 High) $\overline{RD}$ acts as a strobe for both read and write operations. $\overline{WR}$ defines whether a read or write is to be performed (L = write to **NJU25007**; H = there is no provision for reading from **NJU25007**). Since only write operations are allowed, $\overline{WR}$ can be tied low. The recommended sequence to write parallel data to the **NJU25007** in this mode is:

1. Set $\overline{WR}$ low (can be tied to GND).
2. Place data on the data port, D0:7.
3. Assert chip selects, ($\overline{CS0}$ = 0 and CS1 = 1).
   One can be permanently tied to its active state, while the other is controlled.
4. Strobe $\overline{RD}$ high, then low.
5. Deactivate chip selects.
6. Wait at least eight audio samples before sending another command.

Serial Interface

In serial interface mode ($\overline{SEN}$ = 0), D0 is used for the serial data input. Two types of interface modes are supported. In Z80 mode (SEL68 low) $\overline{WR}$ is used as a serial bit clock. In 68K mode (SEL68 high), $\overline{RD}$ is used to clock the serial data. Data is clocked in 8 bits at a time with a maximum data rate of 1 MHz. As in Parallel Mode above, it is recommended that only one command be sent per audio sample while audio is playing. For commands that cause the audio to stop, data can be written to the microprocessor interface at the maximum bit rate of 1MHz (and 1 ms between bytes).

Z80 Mode

When using a microprocessor with a Z80 type interface, writing serial data to the **NJU25007** is done by clocking the data in on rising edges of the $\overline{WR}$, with $\overline{RD}$ held high (SEL68 = 0). The recommended sequence is:

1. Set $\overline{SEN}$ = 0
2. Assert chip selects, ($\overline{CS0}$ = 0 and CS1 = 1).
3. Set $\overline{RD}$ high.
4. Clock in serial data with $\overline{WR}$ (rising edge).
5. Deactivate chip selects.
6. Wait at least eight audio samples before sending next byte (1 ms if audio stopped).

68K Mode

For 68K type microprocessors, the function of $\overline{RD}$ and $\overline{WR}$ is opposite of the Z80. Writing serial data to the **NJU25007** is done by setting $\overline{WR}$ low and clocking the data in on rising edges of $\overline{RD}$ (SEL68 = 1) The recommended sequence is:

1. Set $\overline{SEN}$ = 0.
2. Assert chip selects, ($\overline{CS0}$ = 0 and CS1 = 1).
3. Set $\overline{WR}$ low.
4. Clock in serial data with $\overline{RD}$ (falling edge).
5. Deactivate chip selects.
6. Wait at least eight audio samples before sending next byte (1 ms if audio stopped).

Microcontroller Command Descriptions

NJU25007 Power-Up Default Settings

VMAx	ON (2.1 channel)
Pro Logic	OFF
MxD™	ON (changes to OFF when Pro Logic is selected)
Front Filter	OFF
Subwoofer Crossover	80Hz
Auto Input Balance	ON
Noise Sequencer	OFF
Surround Mode	Pro Logic (only valid in Pro Logic mode)
Surround Delay ON/OFF	ON
Center Channel Mode	ON in WideBand (when Pro Logic is selected changes to ON in Normal)
Level Trim	Master Volume
Trim Level	0dB
Surround Time Delay	20ms
Balance/Tone Select	Balance
Balance/Tone Level	Centered
Subwoofer/Left Surround Swap	SDO1=L/R, SDO2=LS/C, SDO3=SW/RS
Serial Audio Data I/O	18 bits
Serial Audio Data Justification	I^2S
Output L/R Clock	Right when HIGH
Serial Clock for Audio Output	Serial data changes on falling edge
Input L/R Clock Polarity	Left data on HIGH
Serial Clock for Input Audio Data	Input data latches on rising edge
Master/Slave mode	Master
Serial data clock output for converters	32Fs (1/24 DSP clock, 768Fs)
Master clock output for converters	384Fs (1/2 DSP clock, 768Fs)

Configuration Command

After power is applied and a hardware reset is generated on the $\overline{RES}$ pin (pin 30), four bytes must be sent from the microprocessor, which define the hardware settings of the **NJU25007**. This data controls the digital audio interface format, the Clock Generator outputs, and the channel assignment for the output surround data. This command is one of two multi-byte commands—all the others are single byte and take effect immediately without interruption of the audio processing.

When the DSP receives the first byte of the Configuration Command, which is all zeroes, the audio stops and the DSP waits for three more bytes. These hardware settings are intended to be set once, and not changed unless the chip is reset again. The definitions of all the data bits are listed in the subsection entitled "Microcontroller Command Tables" on page 18, with the default settings indicated by *.

Byte #1—Configuration Command opcode, 00h.

Byte #2

SWP	Swap Subwoofer and Left Surround
0	6ch mode: L/R on SDO1, SW/C on SDO2, LS/RS on SDO3
1*	4ch mode: L/R on SDO1, LS/C on SDO2, SW/RS on SDO3* (allows 4ch Pro Logic Surround using only 2 stereo DACs)

BIO	Serial audio data I/O number of bits
0	16 bits
1*	18 bits*

FMT1, 0	Serial Audio Data Justification Format
0, 0	Right-justified
0, 1*	I^2S*
1, 0	Reserved
1, 1	Left-justified

Note: In I^2S Master Mode, LRI becomes an output.

* indicates the default setting

Byte #3—Reserved, 19h

Byte #4—Sampling Frequency Command

LRO	Output L/R Clock
0	Left data on SDOn when High, Right data on SDOn when Low
1*	Right data on SDOn when High, Left data on SDOn when Low*

SCKO	Serial Clock for Output Audio Data
0*	Serial data on SDOn changes on falling edge*
1	Serial data on SDOn changes on rising edge

LRI[1]	Input L/R Clock Polarity
0*	Left data on SDI when High, Right data on SDI when Low*
1	Right data on SDI when High, Left data on SDI when Low

1. In I²S Master mode, LRI becomes an output

SCKI[1]	Serial Clock for Input audio data
0*	Serial data on SDI latches on rising edge*
1	Serial data on SDI latches on falling edge

1. In I²S Master Mode, SCKI becomes an output, and the function is identical to SCKO

MS	Master/Slave
0*	Master mode*
1	Slave mode

ADSCK	A/D-D/A Serial data Clock output
0*	32Fs* (1/24 DSP clock, 768Fs)
1	64Fs (1/12 DSP clock, 768Fs)

ADMCK	A/D-D/A Master Clock output
0*	384Fs* (1/2 DSP clock, 768Fs)
1	256Fs (1/3 DSP clock, 768Fs)

Noise Sequencer Command

A pink noise generator is required for Dolby Pro Logic to set the levels for the front, center, and surround speakers. When the noise sequencer is turned ON, the audio input is ignored, and the noise generator is output on the channel selected, with all other channels muted. The Noise Sequencer works on any of the selected Operating Modes, audible only on the channels in use by the Mode in effect. When the Noise Sequencer is ON, the front High Pass Filters are bypassed.

NS	Noise Sequencer On/Off
0*	OFF*, normal audio output
1	ON, noise generator on selected channel

NCH1, 0	Noise Sequencer Channel
0, 0	Left
0, 1	Center
1, 0	Right
1, 1	Surround (both LS and RS)

Auto Input Balance Command

The **NJU25007** automatically compensates for imbalances in stereo input sources. This is done by continuously comparing the ratio of Center Channel signal to the other signals and minimizing the Center from Left and Right. This assures optimum Surround under varying conditions, such as slight shift in balance due to program changes in TV. This feature may be disabled.

IBL	Auto Input Balance
0	OFF
1*	ON*

Front Filter Command

Smaller front speakers may be used that have limited low frequency response in which case driving them with full bandwidth audio may cause distortion. Turning on the single-pole High Pass Crossover Filters on the Front Left and Right channels will prevent this from occurring. When turned ON, the High Pass -3dB cutoff frequency is set to match the Subwoofer's cutoff frequency of 80Hz. The crossover frequency can be changed using the Coefficient Download Command.

HPF	High Pass Crossover Filter[1]
0*	OFF*, full bandwidth to Front Left and Right
1	ON, 80Hz

1. High Pass Filters are bypassed (full bandwidth) when Noise Sequencer is ON.

Subwoofer Crossover Command

There is a single-pole, low-pass crossover filter for the Subwoofer output with a -3dB cutoff frequency set to 80Hz. When using external filters, the Low Pass Filter on the SW output may be bypassed with full-bandwidth mix of L, C, and R. The crossover frequency may be changed by the Coefficient Download Command.

LPF	Low Pass Crossover Filter
0	Full bandwidth, no filtering
1*	80Hz*

Operating Mode Command

Sets the audio processing to be performed: Dolby Pro Logic, VMAx, Simulated Stereo, or Surround Off (normal stereo).

OP1, OP0	Operating Mode
0, 0	Dolby Pro Logic (5.1ch, L,R,SW,C,LS,RS)
0, 1*	VMAx (2.1ch, L,R,SW)*
1, 0	Simulated Stereo (mono-to-stereo synthesis)
1, 1	Surround OFF (Normal Stereo)

* indicates the default setting

Surround Mode Command

For systems limited to only the front speakers (Left, Center, and Right) Dolby 3 Stereo can be selected which mixes the surround information into the front Left and Right channels. All the Dolby Surround information can be heard, but only through the front speakers. The Center Channel is unchanged between Dolby 3 Stereo and Dolby Pro Logic modes. Phantom Center should never be used with Dolby 3 Stereo mode.

SM	Surround Mode
0	Dolby 3 Stereo (L,C,R, optional SW)
1*	Dolby Pro Logic* (L,C, R,SL,SR, optional SW)

Medianix MxD™ Mono Surround Decorrelation

The surround audio provided by Dolby Pro Logic is monophonic—the same audio data is provided to both the Left Surround and Right Surround outputs. However, the **NJU25007**'s can provide an enhanced surround with simulated stereo from the MxD audio processing. MxD is a proprietary Medianix algorithm that decorrelates the mono signal into Left Surround and Right Surround. In addition to decorrelation, when MxD is ON, the surround channels become full bandwidth, further enhancing realism. In VMAx mode MxD is always on. In Pro Logic operation, MxD defaults to OFF but can be switched ON. When MxD is used with Pro Logic, a separate indicator is required that shows additional processing is operating in addition to Dolby Pro Logic decoding. NOTE: If MxD is selected with 48kHz sampling, the maximum surround delay is reduced from 30ms to 27.5ms.

MxD[1]	Enhanced Surround Decorrelation
0	OFF
1*	ON*

1. **NJU25007** default is VMAx=ON and MxD=ON. When switching to Pro Logic the default for MxD=OFF.

Surround Time Delay ON/OFF Command

Dolby Pro Logic requires a surround delay that is adjustable over a 15 to 30ms range. The **NJU25007** defaults to 20ms delay in both Pro Logic and VMAx operating modes. There may be some cases where zero surround delay is desired. In this case, the delay memory may be bypassed by setting this bit. Certified Dolby equipment may require a front panel indicator showing the departure from the Pro Logic specification. Since the delay memory is 16 bits wide, the output data is reduced to 16 bits when interfacing to 18-bit data input.

DLY	Surround Delay
0*	ON*
1	OFF (bypass)

* indicates the default setting

Center Mode Commands

The Normal Pro Logic Center channel is designed for use with a small speaker compared to the left and right loudspeakers and therefore has a 100Hz high pass filter. Bass signals below 100Hz are redirected to the front left and right channels. If a full range center speaker is used, it may be desirable to reconfigure the **NJU25007** with WideBand Center to allow full bandwidth audio to go to the center channel. Dolby may require an indicator if WideBand is selected when in Pro Logic mode. VMAx always processes full bandwidth center channel information since the small center speaker constraint of Pro Logic is not applicable.

Pro Logic requires a Phantom mode for systems without a center loudspeaker. In Phantom Center mode, the Center audio is mixed equally with the Right and Left front channels to create a center image between the speakers. When operating the **NJU25007** in VMAx mode, this setting has no effect, since Center channel always operates as a phantom. Phantom Center should never be used with Dolby 3 Stereo mode.

Turning the Center Channel OFF has the effect of removing the center image audio, similar to a "Vocal Fader" karaoke mode where the singer's voice or dialog can be diminished. If Center is turned OFF in VMAx, it is not processed into the 2 channel 3D data. However, when operating in VMAx Mode, the Center channel digital audio associated with Pro Logic decoding remains active on SDO2. This allows the possibility of a "virtual 3.1 chanel" configuration if a center speaker is used. In this situation, Center OFF should be selected.

CM1, CM0	Center Mode[1]
0, 0	Normal Center ON (not avalable in VMAx)
0, 1	Phantom Center (not available in VMAx)
1, 0*	Center ON WideBand (full bandwidth)*
1, 1	Center OFF

1. VMAx default is Center=ON, WideBand=ON. Pro Logic default is Center=ON, Normal Center=ON

Level Trim Command

In Pro Logic mode, the Center and Surround Channels are required to have a level trim of +10dB to -10dB. The **NJU25007** provides a range of 0 dB to -31 dB attenuation. To obtain +10dB gain, it is necessary to set the **NJU25007** to nominal -10 dB and apply +10 dB gain in the audio path for that channel. In addition to Center and Surround, the Subwoofer and Master Volume may be attenuated by the same range. Master Volume may be used either as a volume control or headroom control.

The volume and trim control may be done either digitally in the **NJU25007** or with analog techniques. In order to preserve the maximum signal-to-noise audio performance, it is preferable to use analog attenuators external to the **NJU25007** since this preserves the signal to noise ratio (SNR). Digital attenuation only changes the signal's amplitude while the system's noise level remains constant. In effect, the SNR of the system increases as attenuation is increased—the exact opposite of what is desired. Nothing is lost in the **NJU25007** that has over a 140 dB dynamic range. However, system signal-to-noise is limited by the resolution and S/N of the A/D and D/A converters in a digital audio system. After deducting operating headroom for input signal range and level trims, inadequate dynamic range remains unless high resolution data converters are employed. For optimum signal-to-noise performance, the clipping indicator can be used like a VU meter to adjust the input attenuation for maximum signal without clipping.

Level Trims for Center and Surround are active only in Pro Logic and VMAx modes while the Subwoofer is ON in any operating mode that has Level Trim control. In VMAx mode, Level Trim effects both the center and surround audio played through the front loudspeakers (SDO1) along with the Center channel digital audio associated with Pro Logic decoding still active on SDO2.

CH1, CH0	Channel
0, 0	Center
0, 1	Surround
1, 0	Subwoofer
1, 1	Master Volume

TR4-TR0	Trim Level
00000*	0dB*
00001	-1dB
•	•
11110	-30dB
11111	MUTE

* indicates the default setting

Surround Delay Time Command

Dolby Pro Logic requires the surround channels to be delayed from 15ms to 30ms. An internal memory holds 1,536 audio samples. The delay memory can be bypassed using the DLY bit in the Surround Mode Command. When VMAx is ON and the sample rate is at 48kHz, the maximum delay is 27.5ms. Programming 29ms or 30ms will default to 27.5ms only under this condition.

DLY3-DLY0	Delay
1111	30 ms
1110	29 ms
•	•
0101*	20 ms*
•	•
0001	16 ms
0000	15 ms

Balance/Tone Select Command

Balance, Bass, and Treble can be adjusted over a -16dB to +15dB range using the Balance/Tone Level Command preceded by the Balance/Tone Select Command. The procedure is to first send which function is to be adjusted (see table), then send the Level in the following command. The Balance/Tone Control Command only needs to be sent once, since the Balance/Tone Level Command applies to the function previously selected. If no Balance/Tone Control Command has been sent since Reset, the Level defaults to Balance Control. It is acceptable to operate these commands as a pair for each increment. It is recommended to maintain at least 5 ms between the Control and Level Commands. Bass and Treble Tone Controls are implemented using 1st order low-pass and high-pass filters on Left and Right Outputs.

BT1, BT0	Balance/Tone Select
0, 0*	Balance Adjust*
0, 1	Bass Adjust
1, 0	Treble Adjust
1, 1	Reserved

Balance/Tone Level Command

Left/Right balance control is accomplished by first sending the Balance/Tone Control Command followed by the Balance/Tone Level Command. The Balance Level attenuates one channel at a time up to 16dB in 1dB steps according to the following command data. Balance operates in any Operating Mode, but only on the front channels. Left and Right Surround channels are unaffected.

BAL4-BAL0	Left	Right	Balance
11111	0dB	-15dB	Minimum Right
11110	0dB	-14dB	
•	•	•	•
10001	0dB	-1dB	
10000*	0dB	0dB	Center*
01111	-1dB	0dB	
•	•	•	•
00001	-15dB	0dB	
00000	-16dB	0dB	Minimum Left

Tone Control is accomplished by selecting either Bass or Treble Adjust with the Balance/Tone Control Command followed by the Balance/Tone Level Command. The Balance/Tone Level Command affects the Control that was last selected. Bass and Treble are first order low-pass and high-pass filters, respectively, independently adjustable with up to +15dB boost or -16dB cut in 1dB steps. Whenever either control is in boost mode (Level > 0dB), care must be taken to avoid clipping either using the internal Master Volume attenuation or external gain settings. The clipping indicator will activate whenever any output data reaches maximum full-scale level.

The Bass low pass filter cutoff frequency is set at 100Hz, and Treble high pass cutoff frequency is set at 10kHz. Either cutoff frequency may be changed using the Coefficient Download command.

BAL4-BAL0	Bass/Treble
01111	+15dB
01110	+14dB
•	•
00001	+1dB
00000*	0dB*
11111	-1dB
•	•
10001	-15dB
10000	-16dB

Coefficient Download Command

Several default parameters in the **NJU25007** can be changed by downloading a new set of coefficients: Sampling Frequency; VMAx Coefficients for Speaker Angle and 3D Headphone; Bass Low Pass and Treble High Pass Filter Cutoff Frequencies; Subwoofer Cutoff Frequency; Front Left/Right High Pass Filter Cutoff Frequency. The Bass low pass filter cutoff frequency is set at 100Hz, and Treble high pass cutoff frequency is set at 10kHz. The download sequence and the exact data for each is available in a seperate Application Note from NJRC

Microcontroller Command Tables

In the following tables, * indicates the default setting.

Configuration Command (4 bytes)

Byte 1—Download Command

7	6	5	4	3	2	1	0	Byte 1
0	0	0	0	0	0	0	0	Command

Byte 2—Download Data, Hardware Settings

7	6	5	4	3	2	1	0	Byte 2	
SWP	0	0	0	0	BIO	FMT1	FMT0	Data, Hardware Settings	
						0	0	Right-Justified	Audio Data Interface Mode
						0	1	I^2S*	
						1	0	Reserved	
						1	1	Left-Justified	
					0			16 bits	Audio Data Width
					1			18 bits*	
								Reserved	
0								6ch configuration (SW/C,LS/RS)	
1								4ch configuration (LS/C,SW/RS)*	

Note: In I^2S Master Mode, SCK1 becomes an output, and the function is identical to SCKO. Also, in I^2S Master Mode, LRI becomes an output

Byte 3—Download Data—Reserved

7	6	5	4	3	2	1	0	Byte 3
0	0	0	1	1	0	0	1	Data - Reserved

Byte 4—Download Data, Clock Outputs

7	6	5	4	3	2	1	0	Byte 4	
LRO	SCKO	LRI	SCKI	0	MS	ADSCK	ADMCK	Data, Clock Outputs	
							0	384Fs*	A/D - D/A MCLK
							1	256Fs	
						0		32 Fs*	A/D - D/A SCLK
						1		64 Fs	
					0			Master*	Audio Data Interface Mode
					1			Slave	
								Reserved	
			0					↑	Data latches on rising edge*
			1					↓	Data latches on falling edge
		0						L R	Left data on L/R Clock Input when high; right if low*
		1						L R	Right data on L/R Clock Input when high; left if low
	0							↓	Data changes on falling edge*
	1							↑	Data changes on rising edge
0								L R	Left data on L/R Clock Output when high; right if low
1								L R	Right data on L/R Clock Output when high; left if low*

Noise Sequencer

7	6	5	4	3	2	1	0	Byte 1
0	0	1	0	0	NS	NCH1	NCH0	Command
						0	0	Left
						0	1	Center
						1	0	Right
						1	1	Surround
					0			OFF*
					1			ON

Auto Input Balance/Crossover Filters

7	6	5	4	3	2	1	0	Byte 1	
0	1	0	0	IBL	HPF	0	LPF	Command	
							0	Full BW	Subwoofer Cutoff Frequency
							1	80Hz*	
								Reserved	
					0			OFF*	Front L/R High-pass filter
					1			ON	
				0				OFF	Auto Input Balance
				1				ON*	

Operating Mode

7	6	5	4	3	2	1	0	Byte 1
0	0	0	1	1	OP1	OP0	0	Command
								Reserved
					0	0		Dolby Pro Logic (5.1ch)
					0	1		Virtual Surround (2.1ch)*
					1	0		Simulated Stereo
					1	1		Surround OFF (Normal Stereo)

Surround Mode

7	6	5	4	3	2	1	0	Byte 1	
0	0	1	1	1	SM	MXD	DLY	Command	
							0	ON*	Surround Delay
							1	OFF	
						0		OFF*	Surround Decorrelation
						1		ON	
					0			Dolby 3 Stereo (3.1ch)	
					1			Dolby Pro Logic (5.1ch)*	

Center Mode

7	6	5	4	3	2	1	0	Byte 1
0	0	1	1	0	CM1	CM0	0	Command
								Reserved
					0	0		Center ON*
					0	1		Phantom
					1	0		Wideband (full BW)
					1	1		Center OFF

Level Trim

7	6	5	4	3	2	1	0	Byte 1
1	CH1	CH0	TR4	TR3	TR2	TR1	TR0	Command
			0	0	0	0	0	0 dB* Level
			0	0	0	0	1	-1dB
			0	0	0	1	0	-2dB
			•	•	•	•	•	•
			1	1	1	1	0	-30dB
			1	1	1	1	1	MUTE
	0	0						Center Channel
	0	1						Surround Channel
	1	0						Subwoofer Channel
	1	1						Master Volume (Headroom)

Surround Delay

7	6	5	4	3	2	1	0	Byte 1
0	1	0	1	DLY3	DLY2	DLY1	DLY0	Command
				1	1	1	1	30 ms
				1	1	1	0	29 ms
				•	•	•	•	•
				0	1	0	1	20 ms*
				•	•	•	•	•
				0	0	0	1	16 ms
				0	0	0	0	15 ms

Balance/Tone Select

7	6	5	4	3	2	1	0	Byte 1
0	0	0	0	1	0	BT1	BT0	Command
						0	0	Balance*
						0	1	Bass
						1	0	Treble
						1	1	Reserved

Balance/Tone Level

7	6	5	4	3	2	1	0	Byte 1				
0	1	1	BAL4	BAL3	BAL2	BAL1	BAL0	Command	Left	Right		Bass/Treble
			0	1	1	1	1		0 dB	-15 dB	Min. Right	+15 dB
			0	1	1	1	0		0 dB	-14 dB		+14 dB
			•	•	•	•	•		•	•		•
			0	0	0	0	1		0 dB	-1 dB		+1 dB
			0	0	0	0	0		0 dB	0 dB	Center	0 dB
			1	1	1	1	1		-1 dB	0 dB		-1 dB
			•	•	•	•	•		•	•		•
			1	0	0	0	1		-15 dB	0 dB		-15 dB
			1	0	0	0	0		-16 dB	0 dB	Min. Left	-16 dB