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March 2002

It is the policy of ST Audio to improve products as new technology and newer components become available. ST Audio, therefore reserves the right to change specifications without prior notice.

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<p>Note: To avoid damage of your speakers, always turn down the volume or turn off the power of your amplifier connected to the ST Audio MEDIA 7.1 when you are rebooting or shutting down your PC.</p>
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FCC NOTICE

THIS DEVICE COMPLIES WITH PART 15 OF THE FCC RULES.
OPERATION IS SUBJECT TO THE FOLLOWING TWO CONDITION:
(1) THIS DEVICE MAY NOT CAUSE HARMFUL INTERFERENCE, AND
(2) THIS DEVICE MUST ACCEPT ANY INTERFERENCE RECEIVED,
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- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit difference from that to which the receiver is connected.
- Consult the dealer of an experienced radio/TV technician for help.

NOTE : The manufacturer is not responsible for any radio or TV interference caused by unauthorized modifications to this equipment. Such modifications could void the user' s authority to operate the equipment.

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I. Introduction

Congratulations on your purchase of the MEDIA7.1 by ST Audio. The MEDIA 7.1 is a PCI interface card and External Box to connect phono input compatible hardware such as multichannel recorders, digital mixers, synthesizers, etc. with your PC. With much experience in the audio and PC industry over a long period of time and along with valuable suggestions from our customers, our engineers have developed a product we know you will be satisfied with. We guarantee you uncompromising quality as well as excellent technical and audio properties at an extremely affordable price.

What's in the box?

You should have received the following with your MEDIA 7.1 card:

- ST Audio MEDIA 7.1 PCI audiocard
- ST Audio MEDIA 7.1 external box
- D_SUB cable (15pin D-SUB for linking from audiocard to external box)
- 4pin-3pin cable for analog CD-ROM audio connection
- driver and installation CD
- Bundle CD
- this manual

Functions

The MEDIA 7.1 card from ST Audio installs easily and contains everything you need to form a powerful 8 channel output and support powerful hard recording of phono input and also phantom power recording to your Windows PC. Just insert the ST Audio MEDIA 7.1 card into an available PCI slot and connect it with external. The available analog output allows 8channel with your DVD device without a separate external receiver unit.

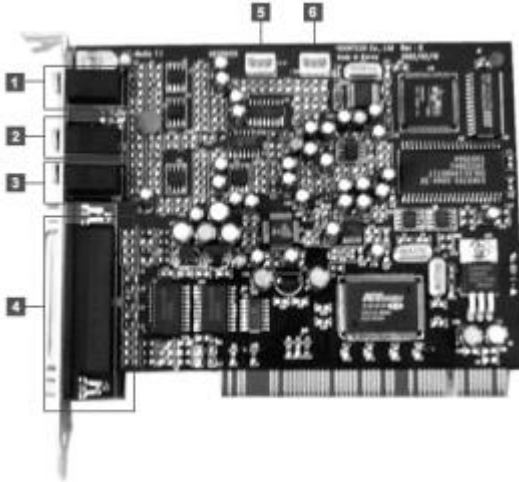
Digital editing with the MEDIA 7.1 card from ST Audio makes many jobs in the studio effortless. For example, using the turn-table you can send audio at up to 24-bit resolution into your computer, graphically edit your tracks in your favorite software application and send them back to tape for mixdown, archiving etc. Another great application is using your digital mixer as the AD/DA audio interface for all famous computer recording software applications.

II. Specifications

- PCI 2.1 compliant interface
- 2 simultaneous analog input channels (24bit/96Khz)
- 8 simultaneous analog output channels (24bit/96Khz)
- AKM4529 ADC/DAC 100dB S/N ratio (a-weighted)
- Optical and Coaxial S/PDIF I/O (24bit/96Khz)
- supported sample rates: 22.05, 32, 44.1 and 48kHz (plus 88.2 and 96kHz for S/PDIF)
- On-board hardware synthesizer (DREAM /Roland GS soundset)
- internal CD in plus Aux In via AC-97 Codec (*Internal Mixer*)
- MPU-401 compatible MIDI In/Out interface
- drivers for Windows 9x/Me, NT4 and Windows 2000 available

III. In- and output connectors

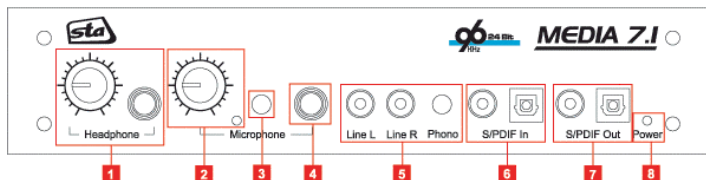
PCI card connectors / options



1. Microphone input to internal device
2. Line input to internal device
3. MO- mix out (connect to MI connector of External Box with stereo cable)
4. D-SUB connects to external box
5. Analog CD input (routed to *Internal Mixer*) – CON1
6. Analog Aux input (routed to *Internal Mixer*) – CON2

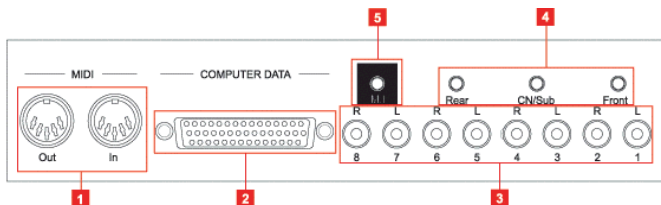
External Box

Font Panel



1. Head phone connector (adjustor and 1/4 inch input connector)
2. Gain controller and peak LED
3. Phantom power (on – phantom on, off – phantom off)
4. 1/4 inch line input connector
5. RCA input connector and Phono input selector (off- phono, on-line input)
6. S/PDIF input connector
7. S/PDIF output connector
8. Power LED

Rear Panel



1. MPU401 MIDI In/Out port
2. D-SUB connector (link D_SUB cable to audio card)
3. RCA analogue output connector
4. Stereo output connector for DVD user
5. MI (mix in) connector – link to MO connector of audio card with stereo cable

IV. Hardware installation

Make sure the computer is switched off and unplugged. Remove the external computer cover. Insert the ST Audio MEDIA 7.1 PCI-card into a free PCI slot making sure that it is pressed firmly into position and secure it to the computer case.

You can use an analog cable to from the CD-ROM drive if you want to listen to audio CDs via the analog monitoring output of the card.

Now install the computer cover again and connect the included D_SUB cable from audio card D_SUB connector to external box of the D-SUB connector.

V. Driver and software installation

Driver installation

The MEDIA 7.1 comes with a CD-ROM that contains drivers for Windows 9x/Me and 2000. The following instructions refer to Windows 98 only. The installation under Windows 95, Windows Me, Windows 2000 and Windows XP is very similar, although the displayed dialog boxes might look different.

After installation of the MEDIA 7.1 in a PCI slot in the computer, boot up the system. Windows will detect the new hardware **automatically** and display the following screen.



Click *Next*.



Select *Search for the best driver for your device* and click *Next*.



Check *Specify a location*. Do not check *CD-ROM drive* or *Floppy disk drives*. Type in <drivename>:\media\driver in the window displayed above and click *Next*. Alternatively, you can click *Browse* and select the correct directory on the CD, then click *Next*.



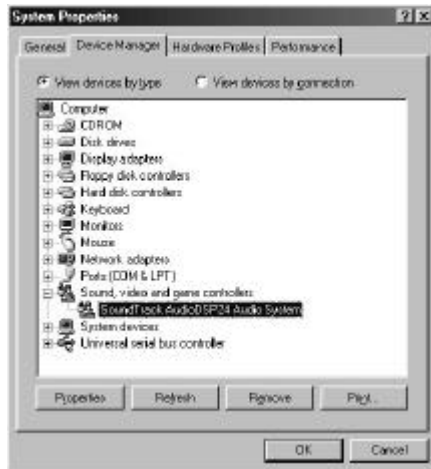
After Windows finds the driver you should see the above screen naming the driver file indicated in the screenshot. Click *Next*.



At this window complete the installation by clicking *Finish*. Windows will now install the driver for you through a series of installation screens. If you are prompted to do anything (this would be unusual), please follow the directions on screen.

After the driver is installed ensure the installation worked by pressing *Start > Settings > Control Panel > System > Device Manager > Sound, video and game*

controllers. You should see the audiocard's device listed there. Highlight it and click on *Properties*. You should see the message *This device is working properly* (this text is not necessarily displayed under Windows 2000). Click on the Resources tab, you should see the message *No conflicts*. If so, congratulations are in order. You have successfully installed the drivers for the ST Audio DSP24 MEDIA 7.1.



Software installation

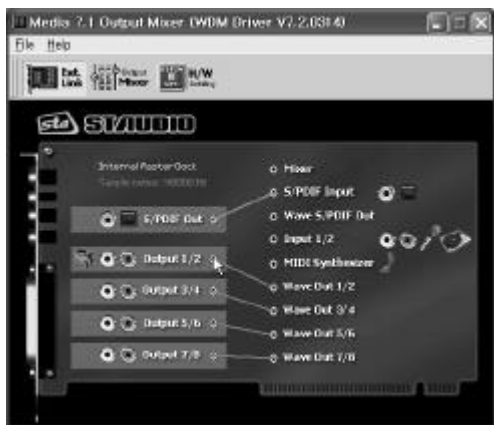
After successfully installing the driver above, Media7.1 output mixer program which control your Media 7.1 card and box is automatically install to your system.

You can find this application program (ST Audio control panel) on "Control Panel"

VI. Media 7.1 Output Mixer

Media 7.1 output mixer is the control center of the MEDIA 7.1 card. It controls the monitoring of your PC based studio. Different studio environments (e.g. analog mixing desk vs. PC as digital mixer) require different monitoring setups. Sometimes you want to listen to the signal you are recording at the moment. Sometimes you want to listen to a modified version of the signal (e.g. processed by an effect processor) or you do want to listen to a completely different audio track while recording. This means that you need to be able to define what exactly is played through the separate output channels of your audiocard and *Media 7.1 Output Mixer* allows you to do that.

What is displayed by Media 7.1 Output Mixer?



The software is divided into two main sections. The output channels of the card are displayed on the left side of the window. The first output channel pair (*Output 1,2*) is used as analog monitoring output with link to *Mixer*. The right section symbolizes the I/O interface on the MEDIA 7.1 card. The connectors on the left side (white circles) are the available sources for the outputs.

The virtual cables for the outputs (which mainly control the monitoring) cannot be removed but replaced by other connections. These connections are displayed as straight lines between the white circles. The available input sources are:

- *Mixer* : Whole available output
- *S/PDIF Input*; the signal from the coaxial S/PDIF input of the card
- *Wave S/PDIF Out*; the playback of the wave device *ADSP24 S/PDIF Out (n)* or channel 9/10 of the ASIO/GSIF driver
- *Input 1,2*; the signal from channel 1/2 of the optical ADAT input
- *MIDI Synthesizer* : Hardware MIDI output from DREAM chipset. (ADSP24 MIDI 2 device)
- *WaveOut 1,2*; the playback of the wave device *ADSP24 Ext. WaveOut 1/2 (n)* or channel 1/2 of the ASIO/GSIF driver
- *WaveOut 3,4*; the playback of the wave device *ADSP24 Ext. WaveOut 3/4 (n)* or channel 3/4 of the ASIO/GSIF driver
- *WaveOut 5,6*; the playback of the wave device *ADSP24 Ext. WaveOut 5/6 (n)* or channel 5/6 of the ASIO/GSIF driver
- *WaveOut 7,8*; the playback of the wave device *ADSP24 Ext. WaveOut 7/8 (n)* or channel 7/8 of the ASIO/GSIF driver



All of these available signal sources can be connected to the outputs on the left side of the *Media7.1 Output Mixer* window.

Output part

For each stereo channel pair it is possible to activate direct monitoring (without any latency). As in the picture, you simply do it by connecting *Input 1,2* to *Output 1,2*. This means that the signal from the input channel 1/2 is sent directly to the output channel 1/2 without any latency / delay. The signal you play inside your application via channel 1/2 is not audible now of course!

If you connect *WaveOut 1,2* with *Output 1,2*, the playback from your audio application is audible via the first output channel pair of the ADSP24 Ext. 1/2 audio device output. The signal from the input is not audible at the same time!



And if you enjoy DVD with your DVD-ROM than you must setting like this. So you can hear each channel of outputs properly.



Mixer

The limitation of these two options for monitoring (which is a limitation of many pro audiocards) is that you cannot hear the input and output signal at the same time. If you use an analog (or digital) mixer that is connected to both the inputs and the outputs of the MEDIA 7.1 card, you can monitor via the mixer so that is no problem. The DSP24 hardware however has a built-in digital mixer so you do not really need an external mixing desk. This function is called *Mixer* (easy to remember!) and as the input and playback signals, you can connect it to *Output 1,2* as shown on the picture.

If the *Mixer* is connected to *Output 1,2* (or alternatively to *S/PDIF Out*), all input and all playback signals are audible simultaneously via the first two channels of the analog monitoring output (or alternatively via the coaxial S/PDIF output). When you open the *Media7.1 Output Mixer*, you can set the volume for each playback and input channel independantly. This allows you to create your own mix completely digitally inside your PC without the need of external mixing equipment and without the need for special software from third parties. This final mix can even be digitally resampled using the *ADSP24 (n) Digital Mixer* MME wave device or channel 11/12 of the ASIO 2.0 driver. Check the section **OUTPUT MIXER** in the following chapter of this manual for reference.

Clock source selection

The clock status display in the upper left corner of *Media7.1 Output Mixer* allow to show present clock status for the MEDIA 7.1 card:

- Internal** Select this (default) if the MEDIA 7.1 should be the master clock source for all operations. All digital devices connected to the S/PDIF output of the card need to work with external synchronisation in this case.
- External** Select this clock source when you want to record from the S/PDIF input with the clock supplied by the device connected to the coaxial digital input of the card. Please note that you cannot select *S/PDIF* as clock source and record from the external digital device like DAT digital input at the same time.

Please note that it is very important that an active device is connected to the digital input (S/PDIF) when the sync is set to external (*S/PDIF*). It is not possible to record a clean signal from the digital input if you use internal synchronisation ... also it is not possible to playback and record clean signals (on any in - and output) if you use external synchronisation without anything connected to the digital input. In fact, on some systems it is not possible to record and playback at all with a wrong sync setting! It is important to use this setting very carefully. It can also be changed in the *Hardware Settings* dialog (acesable via the *File* menu).

VII. Mixer controls

Internal Mixer

Clicking on *Call Internal Mixer* inside *Media7.1 Output Mixer* will launch the *Internal Mixer*: (This mixer is Windows basic Mixer)

The *Internal Mixer* on the MEDIA 7.1 represents basically an additional soundcard which has been 'added' to the audiocard functions. You can use the this section to play Windows sounds or the metronome from a sequenzer, also you could use it for consumer applications such as games or for internet phone usage. Since the *Internal Mixer* even supports DirectSound, you can use it also for some conventional software synthesizer applications. The *Internal Mixer* is exactly same of the Windows mixer (SNDVOL32.EXE - for Windows NT4/2000/XP) is used to control these functions. The *Internal Mixer* is represented by an AC-97 codec chip that is on your MEDIA 7.1 card. AC-97 has been defined as standard by Intel for consumer audio applications on modern PCs. The AC-97 codec ST Audio is using works with a 18bit ADC, a 18bit DAC and an analog mixer. The output of the *Internal Mixer* is audible via the analog monitoring output **only**.

Note that you would never use the *Internal Mixer* for high quality recording or playback for music production. This means: if you have no need for consumer audio applications (as mentioned above) or if you are using a separate consumer soundcard in your PC for such applications, you would never use the *Internal Mixer* and you can stop reading this chapter now. The *Internal Mixer* is a nice addition to the main function set of the MEDIA 7.1 card - not more, not less.

Output Mixer

With the *Output Mixer* you can control all incoming and outgoing signals. Open it by clicking on *Call Output Mixer*.



Note: The *Output Mixer* will only change the audible signal for monitoring purposes. When you are recording an input signal, the *Output Mixer* does not change the signal level. This ensures that you are always recording with best possible audio quality without reduction of the dynamic range.

Master



Controls the output volume of the *Output Mixer*. The signal controlled here will be played through channel 1/2 of output and/or through the S/PDIF output. It is even possible to resample the output signal of the *Output Mixer* internally using the device *ADSP24 Digital Mixer (n)* or channel 11/12 of the ASIO driver.

You can control the the volume of the left and right channels independantly. Muting the signal is possible as well.

Wave (1/2 to 7/8)



In the *Wave* section you can control the volume of the playback signal (for the devices *ADSP24 Ext.(n) WaveOut x/x* or the channels 1~8 of the ASIO and GSIF driver).

Muting each channel is possible. Also you can control the panning independand for the left and right channel of each stereo device.

S/PDIF In / Out



In the *S/PDIF In Out* section you can control the volume of the signal of the device *ADSP24 Ext. S/PDIF In Out (n)* or channel 9/10 of the ASIO and GSIF driver.

Muting the left and right channels is possible independantly.

Input (1/2)



This section is controlling the volume of the different input channels from the analogue input of external box unit.

Muting the left and right channels is possible independantly.

MIDI Sync



This section is controlling the volume of the DREAM MIDI chipset.

If you want to use internal hardware MIDI device than you must choose ADSP MIDI2 device on “Sound and Audio device” of Control Panel.

Notice : ADSP MIDI 1 device is for external MIDI device from external box unit.

VIII. Hardware settings

The Hardware Settings dialog of the MEDIA 7.1 is used to control the basic driver settings of the audiocard. The same dialog is used also for all other DSP24 series cards from ST Audio. You call it via *Media7.1 Output Mixer*. The Hardware Settings dialog is also used when you open the ASIO control panel from an ASIO compatible sequencer such as Emagic Logic Audio or Steinberg Cubase VST.

The upper section allows you to select the card (if you have multiple DSP24 cards installed in your PC) and displays the current version of the driver that is installed. Also you can see the current latency that has been selected for the ASIO 2.0 driver. Below that, you can see the resolution of the converters that are used in your hardware installation. This will be always *24-bit* as the DAC used for monitoring on the card works with 24-bit resolution.

Clock Setting

The *Clock Setting* tab controls the samplerate and bit-depth (resolution) the card operates.

Under *Audio Data Handling* you can determine if the card allows you to playback and record with a bit-depth that is higher than the converter resolution (displayed above). As the converter resolution on the MEDIA 7.1 is 24-bit, it does not matter in most situations what is selected here. The default setting is *Allow operation when data exceeds converter resolution*.

The *Clock and Sample Rates (Hertz)* section allows you to select the source of the clock for the DSP24 card. You can select between *Internal Master Clock* and *External Master Clock*. Make sure to select *External Master Clock* **only** if you will be recording or monitoring from the ADAT or S/PDIF input of the card. In that case, the device that is connected to the input will supply the clock for all operations you perform with the card. If you select *Internal Master Clock* (default), the clock generator on the card will generate the samplerate. You will have problems with noise and clicks in your audio signal if you have selected external



clock without an active device (DAT, MD, digital mixer, etc.) on the digital input. Also you will have similar problems when you want to record a signal from the digital input with the card set to internal clock.

The section of the *Hardware Settings* dialog also shows the current sample rate of the card, which has been set by your application software. The samplerate selected here will be used for all functions of the card. This includes the digital output, so if you sync an external device (MD, DAT, digital mixer, etc.) to the DSP24 card, it will use the clock that is selected here. The *Rate Locked* option is used to force the card to a specific samplerate. It is disabled by default (which is also our recommendation) which automatically allows your application software access to all supported samplerates. When the option is enabled, your software will only work with the selected sample rate. This means that any application that attempts to use the driver at a different sample rate will show an error message. *Reset Rate After Play* should be selected when you want that the sample rate return to a specific setting when no software is using the card for playback or recording. This is especially important if your external device that is connected to the digital output of the card is synched to the DSP24 card. If you just use the analog outputs, the selected samplerate and the *Reset Rate After Play* option are not important when the card is not used by your software at the moment.

Keep in mind that the ADAT protocol does not support higher sample rates as 48kHz. This means that you can only select 88.2 and 96kHz when you are working with S/PDIF signals.

Device Setting

The *Device Setting* tab controls the options needed for playback and recording.

Under *Waveout Driver* you can select the *MME/ASIO* buffer size (latency). The smaller the selected value is, the smaller the resulting latency of the ASIO driver will be (and the other way round). You might notice the latency when you perform some action inside your ASIO application (e.g. you are using a VST instrument in Cubase VST): the result of your action will be audible with a certain delay - the latency. With the current



drivers (v6.x for Windows 9x/Me), the latency can be lowered to 5.5ms at 44.1kHz when the smallest value (256) is selected. The Windows 2000 drivers allow you to lower the latency to 7.5ms at 44.1kHz. If you increase the samplerate, the latency will be automatically lower (e.g. 88.2kHz will give you exactly half of the latency time as 44.1kHz with the same buffer size; keep in mind that ADAT supports 48kHz max.).

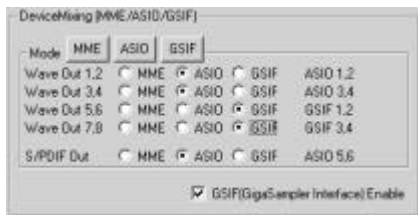
You have to keep in mind that a smaller latency also increases the PC utilization and might have some influence on the performance (very much depending on the PC configuration). A buffer size that is too small may result in clicks or drop outs in the audio signal. Larger buffers could prevent this. This buffer size should be set in the this dialog **before** you launch your music software. In some ASIO applications it is important to re-launch the software even if you have changed the buffer size via the ASIO control panel inside the application. We strongly recommend this for Logic SoundTrack24 and other versions of Logic Audio from Emagic for example but it is also better for all other ASIO applications.

The *WaveDriver* setting under the *MME/ASIO* buffer allows you to change the latency of the MME driver in ms for Windows 2000.

The MME drivers of the DSP24 card automatically synchronize the beginning of recording and playback for all audio devices. For some software applications that are using several MME devices simultaneously, you should select *Single and In-Sync* under *MultiTrack Wave Drivers* to ensure that all channels will begin playback and/or recording at exactly the same time. Otherwise select *Independent*-this allows you to use the cards MME devices with more than one application at the same time.

Multiclient Support (Device Mixing)

The *DeviceMixing (MME / ASIO / GSIF)* section allows you to assign individual stereo output pairs to a certain driver model to be able to use several different programs (e.g. GigaSampler- / Studio and a Audio- / MIDI-Sequencer like Logic Audio or Cubase VST) at the same time. These functions are not available under Windows NT 4.0, 2000 and XP (as GigaSampler-/Studio did not support these operating systems when this manual was created).



Multiclient support is available for MME drivers (e.g. usually used by WaveLab, Sound Forge, Cool Edit, Cakewalk, ...), the ASIO driver (e.g. used by Logic, Cubase VST, Reaktor, ...) and the GSIF driver (used by GigaSampler and GigaStudio). This even allows you to use GSIF software that was designed to run separately on a system without other software parallel to other audio programs.

The example on the picture (previous page) shows a setup where the output channels 1~4 and the S/PDIF output channels are assigned to the ASIO driver. Channels 5~8 are assigned to GSIF. This setup is done **before** you start GigaSampler/-Studio and your ASIO application. It is not possible to change *DeviceMixing* settings after you have launched your audio application(s).

The last column of the section displays the resulting channel numbers. In this example, the S/PDIF output channel (which is usually 9,10) has been assigned to channel 5,6 of the ASIO driver. Also, the channels 5~8 are assigned to channels 1~4 of the GSIF driver. This assignment is needed, because the DSP24 driver simulates a normal audiocard interface in the different driver models for the applications. This prevents compatibility issues with certain audio applications.

It is not recommended to change this setup often, as the different applications will also save the channel assignments internally. If you change the settings in the *DeviceMixing* section, it is wise to double-check the soundcard setup inside the audio apps you are using after that. These options have been invented because of the demand from the users to be able to use GigaSampler/Studio simultaneously to famous Audio-/MIDI-Sequencer software. We strongly recommend to use at least v1.64 of GigaSampler or v2.2 of GigaStudio in order to use the multiclient support. The performance may not be as good when using earlier versions of the software. GigaStudio 2.2 and above will give you best performance.

IX. Usage with software applications

For each application you are using with the MEDIA 7.1 you have to follow some configuration steps. Basically, every stereo input or output channel of the card can be used like a normal soundcard using the corresponding devices. Normally every program will have a configuration function where you can select the devices.

Available in- and output devices

The available wave devices for playback and recording are ("n" will be substituted with the number of the installed DSP24 card, normally 1):

- *ADSP24 Int. WaveOut (n)* – plays audio data through the *Internal Mixer* (which will be audible **only** on the analog output of the card). Please note that this driver only supports 16bit/48kHz as maximum resolution and samplerate.
- *ADSP24 Ext.(n) WaveOut 1/2* – first stereo wave output device. Use this driver to play audio through channels 1/2 of the ADAT output.
- *ADSP24 Ext.(n) WaveOut 3/4* – second stereo wave output device. Use this driver to play audio through channels 3/4 of the ADAT output.
- *ADSP24 Ext.(n) WaveOut 5/6* – third stereo wave output device. Use this driver to play audio through channels 5/6 of the ADAT output.
- *ADSP24 Ext.(n) WaveOut 7/8* – fourth stereo wave output device. Use this driver to play audio through channels 7/8 of the ADAT output.
- *ADSP24 Ext. S/PDIF Out (n)* – plays audio through the stereo S/PDIF digital output.
- *ADSP24 Multiple PCM Out (n)* – this is an interleaved driver for playback of several channels at the same time. Some older audio applications are using this driver to access multiple channels on soundcards.
- *ADSP24 Int. Record (n)* – records the signal from *Internal Mixer*. Use this driver to record from the internal CD or Aux input on the card. Please note that this driver only supports 16bit/48kHz as maximum resolution and samplerate.
- *ADSP24 Ext.(n) Input 1/2* – first stereo input driver to record input signals from your ADAT device (channels 1/2).

- *ADSP24 Ext.(n) Input 3/4* – second stereo input driver to record input signals from your ADAT device (channels 3/4).
- *ADSP24 Ext.(n) Input 5/6* – third stereo input driver to record input signals from your ADAT device (channels 5/6).
- *ADSP24 Ext.(n) Input 7/8* – fourth stereo input driver to record input signals from your ADAT device (channels 7/8).
- *ADSP24 S/PDIF In (n)* – records signal from the S/PDIF input of the card.
- *ADSP24 Multiple PCM In (n)* – this is an interleaved driver to record several channels at the same time. Some older audio applications will use this driver to access multiple channels on soundcards.
- *ADSP24 Digital Mixer (n)* – with this driver you can record the output signal of the *Output Mixer* . This can be used to record output signals from one application in another program for example.

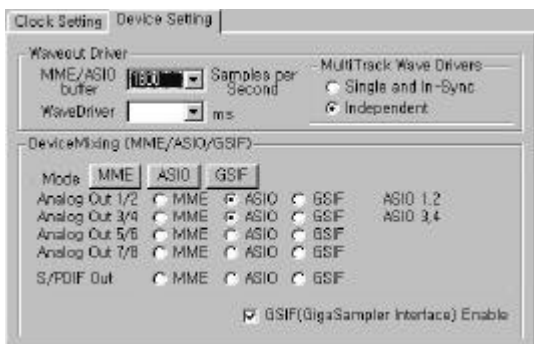
Additionally, the card provides an ASIO 2.0 driver called *ADSP24 ASIO*. It allows access to all 10 input and 10 output channels (8 ADAT + 2 S/PDIF) in applications that support ASIO (like Cubase VST). ASIO allows applications to record and playback at a very low latency (down to 2ms). This means that it is possible to apply realtime effects with your systems CPU input or output signals. Also, you can use software synthesizers to output through ASIO.

Please note that most applications will need special settings to ensure that the MEDIA 7.1 is working properly. We are continuously testing the card with various popular applications. In the following sections we have provided examples of a few applications. Please take the time to review it to avoid problems later.

Setting up Emagic Logic SoundTrack24

Logic SoundTrack24 is a special version of the Logic Software from Emagic, designed to be used with the DSP24 series of audiocards. It can use the ASIO driver of the MEDIA 7.1 for recording and playback. Logic SoundTrack24 works under Windows 9x/Me.

- call *File>Hardware settings* in Media7.1 Output Mixer.



- under *Device Setting* select the ASIO buffer size: we recommend using a value of about 1024 – depending on the system performance.
- click on the *ASIO* button under *DeviceMixing*.
- make sure the option *Rate Locked* under *Clock Setting* is disabled
- set the option *MultiTrack Wave Drivers* to *Independent* (under *Clock Setting*)
- launch Logic SoundTrack24 and select the option *Audio>Audio Hardware & Drivers*.



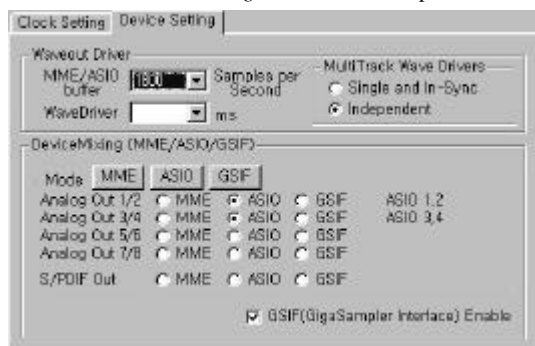
- select *Audio & Hardware Drivers* and select *ASIO ADSP 24* as ASIO device. If you want to work with 24-bit resolution, enable the option *20/24 Bit Recording*.
- exit Logic and launch it again to activate the settings.

Please also refer to the documentation of Logic SoundTrack24.

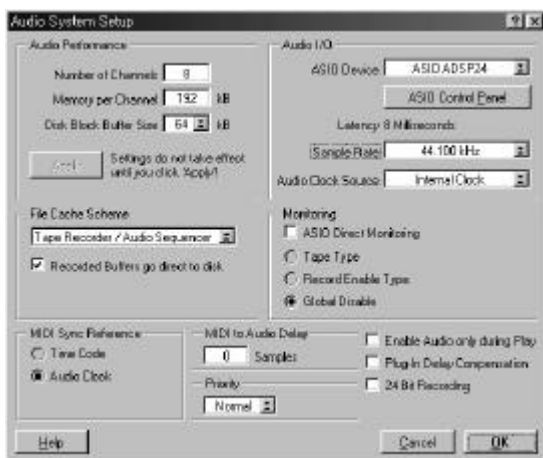
Setting up Steinberg Cubase VST/24 3.72

Cubase VST 3.72 from Steinberg uses the ASIO specification to access recording hardware. Steinberg has defined ASIO and improved support for recording multiple channels at the same time. ASIO features reduced latency for playback of VST plugins and instruments. Please note, that other Steinberg programs (like Nuendo, Cubasis VST or Cubase VST 5.0) have a very similar setup when using the ASIO driver. To setup the MEDIA 7.1 in Cubase VST, please follow these steps:

- call *File>Hardware settings* in Media7.1 Output Mixer.



- Select the ASIO buffer size. On faster systems (Pentium III, Athlon) it is usually safe to use 512 as a small setting. On other systems you may try this setting but increase the buffer size later, depending on the performance. Also you can decrease the buffer size but please test if the performance is OK on your specific system.
- click on the *ASIO* button under *DeviceMixing*.
- Start Cubase VST and select the option *Audio>System*.



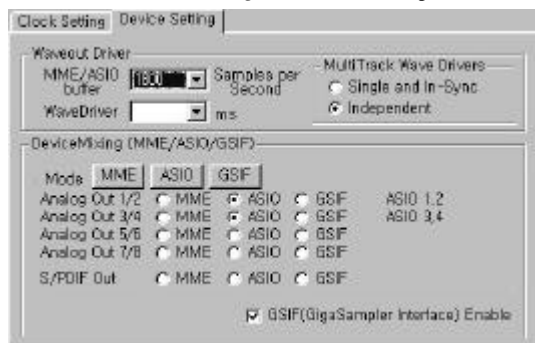
- In the section *Audio Performance*, set *Number of Channels* to 16 (recommended).
- In the *Monitoring* section, select *Global Disable* (recommended). If you want to monitor signals in realtime and adding VST plugin effects at the same time, select *Tape Type* instead.
- In the upper right corner under *ASIO Device* select *ADSP24 ASIO*. By clicking on *ASIO Control Panel*, you can also change the *samples per buffer* value.
- Below that you can choose the *Sample Rate*.
- If you want to use the DSP24's realtime monitoring functions (without adding effects to the monitor signal), select *ASIO Direct Monitoring*

To define which input and outputs to use in Cubase VST, first go to *Audio>Inputs*. A window will appear that allows you to activate or deactivate each input (stereo). You can assign these inputs to the channels in the *VST Channel Mixer* then (via *Audio>Monitors*). For the outputs, you need to open *Audio>Master*. In this window you can deactivate a stereo bus for each stereo output. These busses can be used internally in Cubase VST to route signals to the outputs. You can find more information about this in the Cubase VST documentation. Also, the latest updates for Cubase VST can be downloaded from <http://www.steinberg.de>.

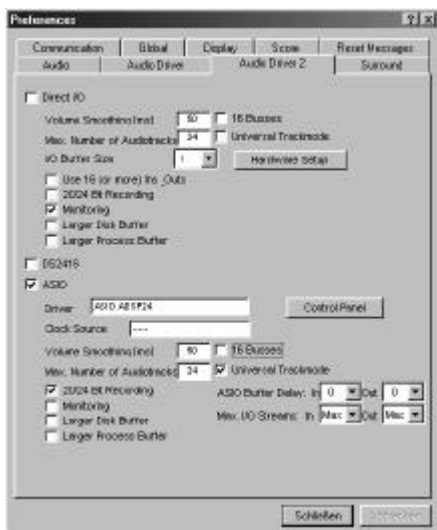
Setting up Emagic Logic Audio Platinum 4.7

The powerful Audio-/MIDI-Sequencer Logic Audio Platinum 4.7 (and higher) from Emagic also uses ASIO drivers. Follow these steps to setup the application for the MEDIA 7.1.

- call *File>Hardware settings* in Media7.1 Output Mixer.



- select the ASIO buffer size: we recommend using a value of about 1024 – depending on the system performance.
- click on the *ASIO* button under *DeviceMixing*.
- launch Logic Audio Platinum and select the option *Audio>Audio Hardware & Drivers>Audio Driver 2*.



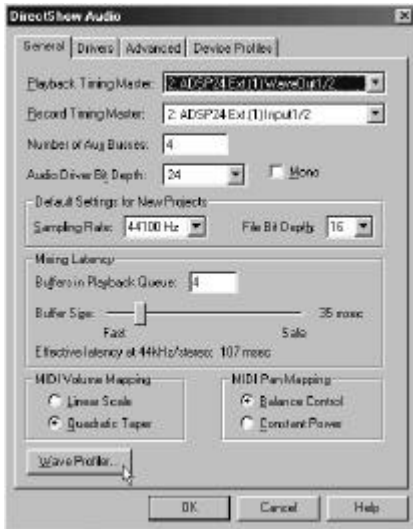
- check the box *ASIO* and select *ASIO ADSP24* as ASIO device. If you want to record with 24-bit resolution, check the box *20/24 Bit Recording*
- exit Logic Audio and launch it again to activate the changes.

You can get the latest informations about Logic Audio Platinum at <http://www.emagic.de>. If you use previous 4.xx versions of Logic Audio, we recommend to get the free update to Version 4.7 at least.

Setting up Cakewalk Pro Audio 9.x

Cakewalk Pro Audio is a famous Audio-/MIDI-Sequencer and recordingsoftware. It uses the MME drivers of the MEDIA 7.1. Other recent versions of Cakewalk need a very similar setup.

- launch Cakewalk and select *Options>Audio*. The *DirectShow Audio* dialog will be displayed.



- Click on *Drivers*. You will see a list of MME devices for Input and Output. Select the entries you want to use (check the section AVAILABLE IN- AND OUTPUT DEVICES earlier in this chapter for reference). Usually you should select at least *ADSP24 Ext. (1) Input 1/2* and *ADSP24 Ext. (1) WaveOut 1/2*.
- Go back to *General* and select *Wave Profiler*. Follow the directions on screen until you see the message *Audio hardware has been successfully profiled*.
- Select *ADSP24 Ext.(1) WaveOut 1,2* as *Playback Timing Master* and *ADSP24 Ext. (1) Input 1,2* as *Record Timing Master*. If you are recording from the S/PDIF input, select *ADSP24 S/PDIF In (1)* instead. The option *Audio Driver Bit Depth* should be set to 24.
- Select the sample rate and resolution you want to use under *Default Settings for New Projects*.
- *Mixing Latency*: Basically, you can set *Buffers in Playback Queue* as 4. Settings less than 4 will result in lower latency but the system performance could be lower, resulting in possible drop-outs. Start with buffer sizes suggested by the *Effective latency* tests, usually around 100msec. When the latency needs to be under 100msec, click on the *Advanced* tab and enable the *WavePipe(TM) Acceleration* option. Please note that each system will have

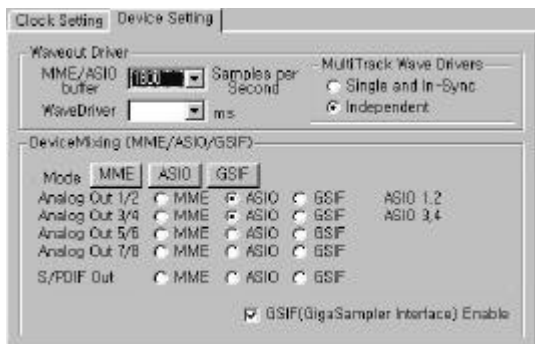
different optimal settings. You should experiment with the settings to get the perfect setup for your system.

- Under *Device Profiles*, enable the option *Use Wave Out Position For Timing*. You can get the latest updates and informations on Cakewalk on <http://www.cakewalk.com>.

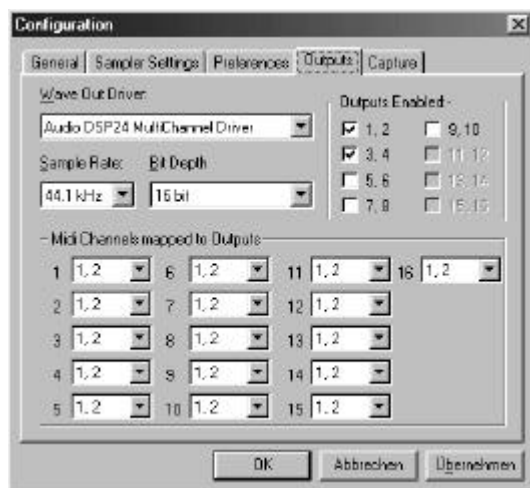
Setting up Nemesys GigaSampler 1.64

GigaSampler from Nemesys is a powerful and very popular software sampler. The MEDIA 7.1 provides a special GSIF-driver (GigaSampler-InterFace) that allows you to use the separate outputs of the card with GigaSampler. We do not recommend to use older versions than GigaSampler 1.64. Also, the performance using the newer GigaStudio 2.2 (and above) by Nemesys is much better.

- call *File>Hardware settings* in Media7.1 Output Mixer.



- press on the *GSIF* button under *DeviceMixing* – or select *GSIF* for **at least** one channel.
- launch GigaSampler and select *Config>Outputs*.



- Under *Wave Out Driver*, select *Audio DSP24 MultiChannel Driver*.
- Select the Output channels you want to use on the right side under *Outputs Enabled*.
- Now assign the *Midi Channels mapped to Outputs* in the lower area.

You can get the latest informations and updates of GigaSampler on their website <http://www.nemesysmusic.com>.

X. Troubleshooting / FAQ

This chapter addresses common questions to solve possible problems and avoid conflict with other PC hardware or specific system configurations.

Q: Can I use the card with an AMD Athlon / Duron / Thunderbird CPU? Will it work with a VIA Cyrix CPU?

A: Yes, of course! There are no compatibility issues with microprocessors. Please also check the following question about VIA chipsets, as many mainboards for AMD and VIA CPUs are based on VIA chipsets.

Q: What about VIA chipsets? Can I use my VIA chipset based mainboard with the card?

A: Yes, you can use VIA based systems with the DSP24 series of cards. Unlike with some other chipsets, the configuration in this case is more 'sensitive' and the system needs to be properly configured to avoid possible trouble:

Please make sure the DSP24 PCI card gets a unique and unshared IRQ. Secondly, a recent version of the VIA 4-in-1 driver (we recommend at least v4.31) and the VIA USB Filter driver must be installed (check <http://www.via.com.tw/> or the website of your mainboard manufacturer). If you are using Windows 98SE with USB enabled, Microsoft's UHCD Update should be used as well (do a search for 'uhcd.sys' on Microsoft's Knowledge Base).

Please check if there is an update for the BIOS of your mainboard ... if so, please download and use it.

On mainboards with the KT133 and KT133A chipset, you must use high quality 133MHz memory modules only. Also check the corresponding setting in your BIOS (it should be set to 133MHz) and make sure the memory timing is not set to an incorrect (e.g. too low) value.

Set PCI DELAY TRANSACTION in your BIOS to DISABLED. Then, set PCI DEVICE LATENCY TIMER to a value of 32 or lower. PCI MASTER READ CACHING should be set to DISABLED as well (could cause performance problems, but because of the VIA southbridge bug, this setting is required).

Last but not least, we recommend to disable all power management related functions.

Q: Is it possible to install this card in a system with onboard-sound or another soundcard?

A: Usually yes. As long as the other soundchip does not use a Envy24 / ICE1712, there should be no problems. You should make sure that both devices do not share their IRQ.

Q: How many available IRQs do I need to use the card?

A: Exactly one. It should be unshared if possible, because the DSP24 is a busmaster device ... otherwise you might get performance problems. It does not matter which IRQ number has been assigned to the card.

Q: I have read that the card is full duplex, but I can't do simultaneous record and playback with Cakewalk!

A: Simultaneous record and playback needs to be enabled in Cakewalk Pro Audio. In the software, go to *Settings > Audio Options > Advanced* and make sure *Enable simultaneous record and playback* is checked.

Q: I am using Cubase VST 5.0 and I'm getting crazy: always when I change a setting inside Media7.1 Output Mixer or the mixer, the playback suddenly stops!

A: Make sure the menu entry *Options > Active in Background* is checked inside Cubase. This is a general Cubase setting and not related to the audiocards driver.

Q: I am using Cubase VST 3.xx and I'm getting crazy: always when I change a setting inside Media7.1 Output Mixer or the mixer, the playback suddenly stops!

A: Goto *File > Preferences* and make sure that *Active in Background* is checked for *MIDI and Audio*. This is a general Cubase setting and not related to the audiocards driver.

Q: When recording or playing audio, I get an audible click about every 3 to 5 seconds. Is the hardware defective?

A: No! First of all, check the following two settings:

Make sure the 'auto-play' function for CD-ROMs is disabled. Goto *Control Panel > System > Device Manager > CD-ROM* and select *Properties* for all CD-ROM drives. Now disable *Auto Insert Notification* under *Settings*.

DMA access for your IDE HDDs must be enabled. Goto *Control Panel > System > Device Manager > Disk Drives* and select *Properties* for each IDE-HDD you have installed (an entry is typically called *GENERIC IDE DISK TYPExx*). Now enable *DMA* under *Settings*.

Please check these settings from time to time. Some Windows installations 'loose' the DMA and CD-ROM notification settings when you are changing certain elements in the hardware configuration or if you boot Windows in Safe Mode.

Q: I can hear cracks and noise when I play my recorded audio signal! Is this a hardware problem?

A: In most cases this problem can be solved by changing the interrupt which has been assigned to your hardware. Check in *Control Panel > System > Device Manager > Computer > Properties* if there is another hardware component using the same interrupt as the DSP24 PCI-card (it should have its own interrupt, especially on mainboards using the VIA chipset). Except an *IRQ-Holder* entry, no other device should be listed with the same IRQ number. If you find such a problem, you usually have to install the DSP24 PCI card into a different PCI slot. On some mainboards you can change the IRQ assignment inside the BIOS of your mainboard.

Q: I can hear noise when I move a window or fader on screen inside my audio application. I am using an ATI Rage 128 graphic card.

A: Please download and install current drivers for your graphic card and set the bit-depth to 32bit (not lower). This will reduce the system load the graphic card is causing.

Q: I can hear noise when I move a window or fader on screen inside my audio application. I am using a Matrox Millenium/Mystique/G200 card.

A: Please download and install current drivers for your graphic card. Inside the Matrox specific options you can find a setting called *Busmastering* - please disable it.

Q: Can I use Logic SoundTrack24 under Windows 2000/XP?

A: No, this is not possible. All Logic 4.x versions from Emagic do only work under Windows 9x/Me ... Windows 98 SE is recommended.

Q: I am not able to change the MME / ASIO / GSIF settings for the multicient driver in the Hardware Settings dialog.

A: Under Windows NT 4.0 and Windows 2000, these settings are not available as there is no multicient support for GigaSampler-/Studio on this operating system. GigaSampler-/Studio currently do not work under Windows NT 4.0 or Windows 2000 (that is not related to our driver, that is a general limitation).

Q: Why do I get error messages that the ASIO Multimedia test/configuration failed using Cubase VST, Cubasis VST or Nuendo from Steinberg?

A: This test is done by the ASIO Multimedia driver from Steinberg that simulates ASIO for soundcards without ASIO support. It would disappear as soon as this driver is configured properly. However, as the DSP24 series of cards have their own ASIO driver for Windows 9x/Me/2000, you don't need to run this test at all. Just select the ASIO driver (*ASIO ADSP24*) and then the ASIO Multimedia driver will no longer be used.

XI. Support / Contact

ST Audio offers various resources to provide customer support. Please visit our website **www.STAUDIO.com** to access our customer service area.

The Q&A forum on the website provides help – please use it to discuss all issues related to our products. Our qualified support staff from all over the planet (and some from other planets) will help you to solve issues you may have. Do not hesitate to use the Q&A forum for **any** question you might have. Other customers read the messages and will possibly reply to the requests on this forum as well.

You can find updated versions of drivers, applications and documentation on our webserver.

The extensive Knowledge Base and FAQ sections of our website, deal with common issues and are updated frequently by us. Please check this resource from time to time to get the latest updates on your DSP24 card.