marian
D I G I T A L   A U D I O   E L E C T R O N I C S

MARC A

User's Guide

© April 2007, Marian v2.3

Hardware Design by Marian
"SyncBus" developed by Marian
User's Guide by Wilke Pfannkuch – Hamburg, Germany

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Welcome

thank you very much for your decision to purchase one of the products from the MARC product series.

Please consider taking some time to study this guide. Besides traditional installation and operation instructions it contains information, that will make the handling with the MARC A easier for you.

We wish you lots of fun and success in working with the MARC A.

1. Features

Your MARC A provides some special characteristics. In the following you will get a short overview of them:

- 1 ADAT™ Input (8 Channels)
- 1 ADAT™ Output (8 Channels)
- Support of 24 bit ADAT™ audio transfer
- Simultaneous operation as playback clock master and recording clock slave
- Multiple card support (up to 4 cards can be installed)

2. Scope of Supply

First of all, please check, whether you received the following components undamaged and complete.

- MARC A PCI card
- User's Guide
- CD-ROM incl. drivers

3. System Requirements

In order to be able to operate the MARC A, your PC should meet the following requirements:

- AT compatible PC with a spare PCI slot
- Pentium or AMD processor
- Windows™ 95/98/ME or Windows™ NT4 SP3 or Windows™ 2000/XP
- Microsoft DirectX 9

Please note, that your used audio software may demand different requirements.

4.   Ports

4.1.   External Ports

The picture on the right shows the slot bracket of your MARC A with its external audio ports.
For the connection of external devices to the ADAT interface you require optical connection, which you can purchase in specialist shops.

When using optical cables, please ensure that they are free of breaks and damage, because this may lead to failures.

ADAT OUT

ADAT IN
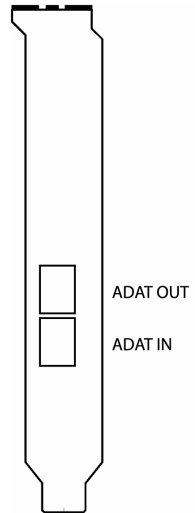
4.2.   Internal Ports

The picture on the right shows the board of the MARC A with its internal ports.
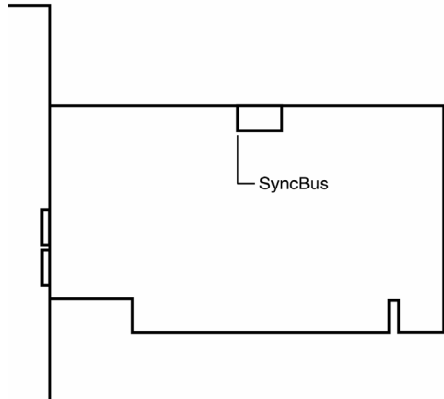
SyncBus

If you use several MARC cards or other cards developed by Marian, you can connect the SyncBus ports of the individual cards using a SyncBus cable which is available as accessory. You can order/purchase this cable from your dealer.

For synchronous recording and playback with several devices a shared clock is required. The SyncBus takes care that in such a situation all recording and playback devices synchronized by the SyncBus are started and stopped sample-accurately and simultaneously. It is not relevant whether the recording and playback devices are on the same or on different cards. For details on the settings see chapter 6.2.1.1. ff.

5.   Hardware Installation

Following the described steps below you can install up to four cards of the MARC A on the same PC.

1. Turn off your PC and all devices connected to it, disconnect the power supply.
2. Open the case.
3. Remove the MARC A from the anti-static foil. Make sure to hold the card only at its edges or the slot brackets. Do not touch any components or contacts.
4. Insert the card carefully and rectangular into a spare PCI slot. Ensure that the card was inserted properly into the slot.
5. Screw the card on the slot bracket to the case.
6. Close the PC case and reconnect it to the power supply.

6.   Driver Installation

6.1.   About MME, DirectX, WDM-Audio, ASIO and GSIF

You can use the Marc A with many audio software applications. The Marc A driver software with all its interfaces makes that possible. An audio application uses a driver interface to transfer the audio data to and from the Marc A hardware. In many audio applications you are able to setup a specific driver interface to be used.

For connecting software and hardware you can use the driver interfaces MME, DirectX or DirectSound, WDM-Audio, ASIO 2.0 and the Tascam GigaSampler Interface (GSIF). WDM-Audio can be used only with Windows™ 2000/XP.

You can configure several driver interface according your personal needs. Please refer chapter "7.1.2 Audio Settings on Windows™ 98/ME" and "7.1.3 Audio Settings on Windows™ 2000/XP".

If you use an ASIO compatible audio software, you will achieve extremely higher performance of the system as well as shorter latency periods. In an ideal case 2 ms are possible!

6.2. Installation on Windows 95 (Release 950 and 95a)

1. Start your PC after installing your hardware.
2. Windows automatically recognizes at startup the newly installed device and starts the Hardware Wizard.
3. The *"A new hardware component was found"* window appears.
4. When selecting the driver which is to be installed, please select the option *"Driver from disk provided by hardware manufacturer"* and confirm your selection.
5. Insert the supplied CD in your CD-ROM drive.
6. Select your CD-ROM drive using the "*Browse*" button, change to *\marca\win9x\english* directory and confirm your selection. Windows copies the driver files and installs MARC A.
7. After finishing the installation, your MARC A is ready to be operated without having to restart the system.

6.3. Installation on Windows 95 (Release 95b and 95c)

1. Start your PC after installing your hardware.
2. Windows automatically recognizes at startup the newly installed device and starts the Hardware Wizard.
3. The *Wizard for device driver updates* appears.
4. Insert the supplied CD in your CD-ROM drive and click *"Next"*.
5. Windows now searches for the updated drivers, but it cannot find any. Click *"Other position"* and click "*Browse*" in the opening window. Change into the *\marca\win9x\english* directory on the CD and confirm it by clicking the *"OK"* button.
6. Windows copies the driver files and installs the MARC A. During this process you are asked again for the driver's directory. Select the directory as described previously in step 5.
7. After finishing the installation, your MARC A is ready to be operated without having to restart the system.

6.4. Installation on Windows 98

1. Start your PC after installing your hardware.
2. Windows automatically recognizes at startup the newly installed device and starts the Hardware Wizard.
3. During the start of Windows the Hardware Wizard searches for new drivers. Click *"Next"* to start the search.
4. Enable the *"Search for the best driver for your device"* option and confirm by clicking *"Next"*.
5. Insert the supplied CD in your CD-ROM drive.
6. Select your CD-ROM drive in the next dialog, change into the *\marca\win9x\english* directory and then click *"Next"*. The

Hardware Wizard searches the CD-ROM for the appropriate drivers and finally displays the names of the drivers.

7. Click *"Next"*; Windows copies the driver files and installs the MARC A.

8. After finishing the installation click *"Finish"*. The driver will be enabled and the MARC A is ready to be operated without having to restart the system.

6.5. Windows ME

1. Start your PC after installing the hardware.

2. At startup Windows automatically recognizes the newly installed device and starts the Hardware Wizard.

3. Insert the supplied CD in your CD-ROM drive.

4. Enable the *"Search for the best driver for your device"* option and confirm by clicking *"Next"*.

5. The Windows Hardware Wizard now searches for the best drivers and finds all Marc A drivers on the CD. Choose the drivers for Windows 95/98/ME (folder *"\marca\win9x"*) and your preferred language. Confirm this dialog with *„Next"*

6. Windows copies the driver files and installs the MARC A.

7. After finishing the installation click *"Finish"*. The driver will be enabled and the MARC A is ready to be operated without having to restart the system.

6.6. Installation on Windows NT 4

1. Start your PC after installing your hardware.

2. Log in using administrator rights.

3. Open "*System Control*" and choose *"Multimedia"*, then change to the *"Devices"* tab.

4. Click *"Add"* button and confirm the opening window by clicking *"OK"*.

5. Insert the supplied CD in your CD-ROM drive.

6. The *"Install driver"* window opens. Click "*Browse*", select your CD-ROM drive and on the CD-ROM select the *\marca\nt4\english* directory.

7. Confirm your selection by clicking *"OK"*. Windows shows the recognized drivers in the *"Unlisted or Updated Driver"* window.

8. Confirm the window by clicking *"OK"*. The driver files are then copied.

9. Then the MARC A is available without having to restart the system.

6.7. Installation on Windows 2000

1. Start your PC after installing your hardware.
2. Log in using administrator rights.
3. Windows automatically recognizes at startup the newly installed device and starts the Hardware Wizard.
4. During the start of Windows the Hardware Wizard searches for new drivers. Click *"Next"* to start the search.
5. Click *"Next"* in the following *"Install hardware drivers"* window.
6. Insert the supplied CD in your CD-ROM drive.
7. Confirm the *"Browse for driver files"* dialog by clicking *"Next"*.
8. The Hardware Wizard automatically browses the CD for suitable drivers and finally displays their names.
9. Click *"Next".* Windows copies the driver files and installs the MARC A.
10. At this stage Windows prompts you whether you'd like to continue the installation without an existing digital signature. Please click *"Yes"*.
11. After finishing the installation click *"Finish".* The drivers will be enabled and the MARC A is ready to be operated without having to restart the system.

6.8. Installation on Windows XP

1. Start your PC after installing your hardware.
2. Log in using administrator rights.
3. Insert the supplied CD in your CD-ROM drive.
4. Windows automatically recognizes at startup the newly installed device and starts the Hardware Wizard.
5. Enable the option „Install Software automatically" and confirm by clicking „Next".
6. Ignore the next message and click „Continue"
7. Windows copies the driver files and installs the MARC A.
8. After finishing the installation the MARC A is ready to be operated without having to restart the system.

7. MARC A Manager

After successful driver installation the symbol shown on the right appears for the MARC A Manager in the Windows task bar.
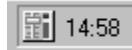
	Windows™ 98/ME You open the MARC-Manager by double-clicking the taskbar symbol. The dialog box with checkboxes is then displayed.
	Windows™ 2000/XP With a simple mouse-click you can open a menu, in which you can select what kind of settings you want to change.
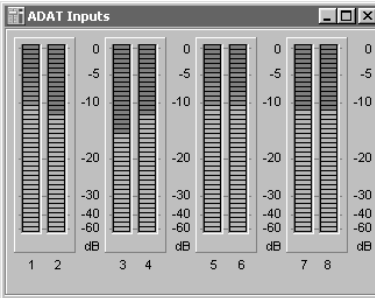
"ADAT Inputs"/"Inputs" opens a windows where you can control the signal levels of the ADAT™ input channels.
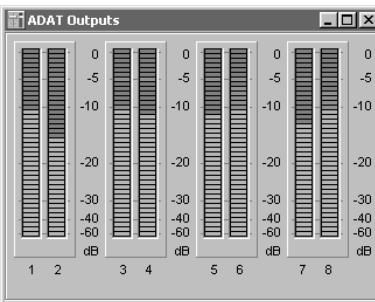
"ADAT Outputs"/"Outputs" opens a windows where you can control the signal levels of the ADAT™ output channels.

"Settings" opens a dialog box, in which you can change the settings for clock and synchronization, DirectSound and GSIF.

„About" opens a window with version and contact information.

„Exit" ends the execution of the MARC A Manager software.

7.1. Settings

7.1.1. "Synchronization" Tab
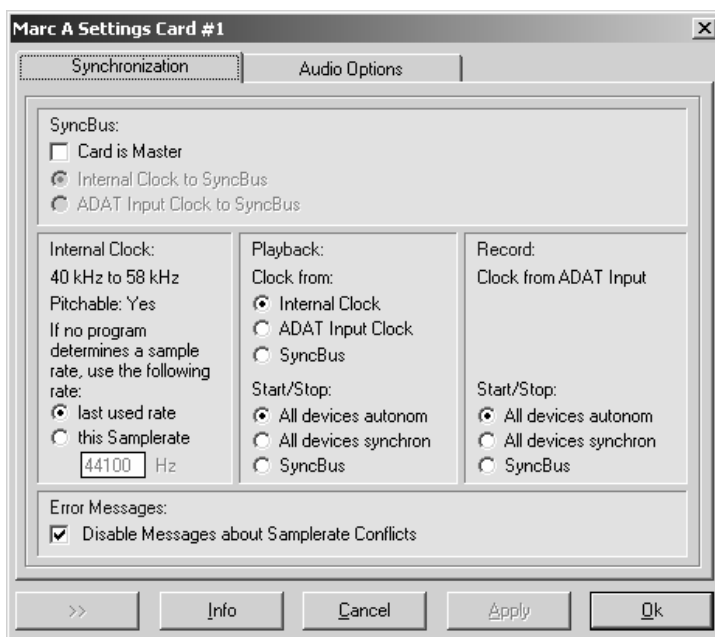
In this window you perform synchronization settings for the MARC A.

Please note that these settings do not influence the function of the audio applications which use ASIO or GSIF driver interface. In this case the settings are determined by the audio application or automatically.

What's a clock?

The clock is a measure which directs all digital devices in a digital device environment. The clock is necessary for the synchronous operation and the data transfer of the digital devices. If the devices are directed by the same clock, they work simultaneously. If there is no clock for a device, you cannot perform any recording or playback on this device.

7.1.1.1. "SyncBus" Section

"Card is master"

means that MARC A is a clock master and is responsible to provide the clock on the SyncBus.

If several cards are to be synchronized using the SyncBus, exactly one card has to be configured as master. For all other cards the "Card is master" option needs to be disabled.

"Internal clock to SyncBus"

If this field is active, the sample rate of the internal clock generator is transferred to the SyncBus. It will be used by the cards, which are configured as SyncBus slaves, as clock reference.

"ADAT input clock to SyncBus"

means that the clock of the signal at the ADAT input of the MARC A is transferred to the SyncBus.

Please note that all devices synchronized via SyncBus will not function properly if there is no valid signal at the ADAT input of the MARC A.

7.1.1.2. "Internal Clock" Section

In this section you can see the properties of the internal clock generator. Further you can determine, which sample rate is generated by the clock generator, if the sample rate is not given by any audio application.

In order to prevent difficulties when synchronizing with other devices, we recommend always to operate at 44.1 kHz or 48 kHz.

7.1.1.3. "Playback" Section

In this section you configure the playback part of the MARC A.

In the *"Clock from"* section you determine how the playback sample rate is set.

"Internal Clock"

When choosing this setting, the playback sample rate is generated by the internal clock generator and is transferred to the ADAT output of the MARC A. The sample rate is either given by an audio application or – in case that there is no audio application using the MARC A playback at this moment – it is determined by the given sample rate. Externally connected devices receive this sample rate at their appropriate inputs.

The playback runs in its sample rate totally independently from the sample rate of the SyncBus or the input signal.

"ADAT input clock"

The sample rate for playback is determined by the clock of the input signal. This way recording and playback are operating absolutely synchronously concerning their sample rate. You must ensure that the device, which is connected at the input, generates a clock. That is, it has to be configured as clock master. In this mode the sample rate cannot be influenced by any audio software. Playback again operates regarding its sample rate independently from to the sample rate on the SyncBus.

Please note that all devices synchronized via "ADAT Input" will not function properly if there is no valid signal at the ADAT input of the MARC A.

"SyncBus"

The sample rate for playback is determined by the sample rate on the SyncBus. Because of this playback operates synchronously to all devices which are connected by the SyncBus.

In this mode the playback sample rate of the MARC A cannot be influenced by any audio software. This will have to be done by the controlling a device on the clock master card.

Please note that all devices synchronized via SyncBus will not function properly if there is no clock provided.

Playback operates regarding its sample rate independently from to the sample rate of the input signal.

In the *"Start / Stop"* section you can set the behavior of the individual devices when starting and stopping a recording or playback session. These settings are not related to the clock settings.

"All devices autonomously"

determines that playback devices can be started and stopped independently. This setting could be useful when several audio application have to use several outputs.

"All devices synchronously"

determines that all playback devices are started/stopped sample-accurately and synchronously. This setting could be useful when you use for one multi-channel playback all playback devices of the MARC A and you'd like to operate the playback devices independently from the recording devices.

On Windows™ 2000/XP there are some restrictions using this option. Please refer chapter "7.1.3.1 WDM, MME and DirectSound"

"SyncBus"

determines that all playback devices connected by the SyncBus are started/stopped sample-accurately and synchronously. This setting could be useful when playback is to be started/stopped synchronously with the recording, or the playback devices of the MARC A are part of a multi-channel hard-disk recording project and are to be synchronized for the start/stop of other MARIAN sound systems.

On Windows™ 2000/XP there are some restrictions using this option. Please refer chapter "7.1.3.1 WDM, MME and DirectSound"

7.1.1.4. "Recording" Section

"Clock from ADAT input"

The clock for the recording part of the MARC A cannot be determined. The sample rate for an ADAT™ recording will always be given by the clock of the input signal.

In the *"Start / Stop"* section you will find options which are explained in the following:

"All devices autonomously"

Using this option all recording devices can be started/stopped independently from each other. This setting could be useful when several audio applications performs recordings separately using the MARC A.

"*All devices synchronously"*

Using this option all recording devices can be started/stopped sample-accurately and synchronously. This setting could be useful when you'd like to use all recording devices of the MARC A for a multi-channel recording, but you intend to operate the playback devices independently from this.

On Windows™ 2000/XP there are some restrictions using this option. Please refer chapter "7.1.3.1 WDM, MME and DirectSound"

"SyncBus"

Using this option all recording devices together with all devices connected by the SyncBus can be started/stopped sample-accurately and synchronously. This setting could be useful when recording is to be started/stopped synchronously to playback, or the recording device of the MARC A is part of a multi-channel hard-disk recording project and is to be synchronized for the start/stop of other MARIAN sound systems.

On Windows™ 2000/XP there are some restrictions using this option. Please refer chapter "7.1.3.1 WDM, MME and DirectSound"

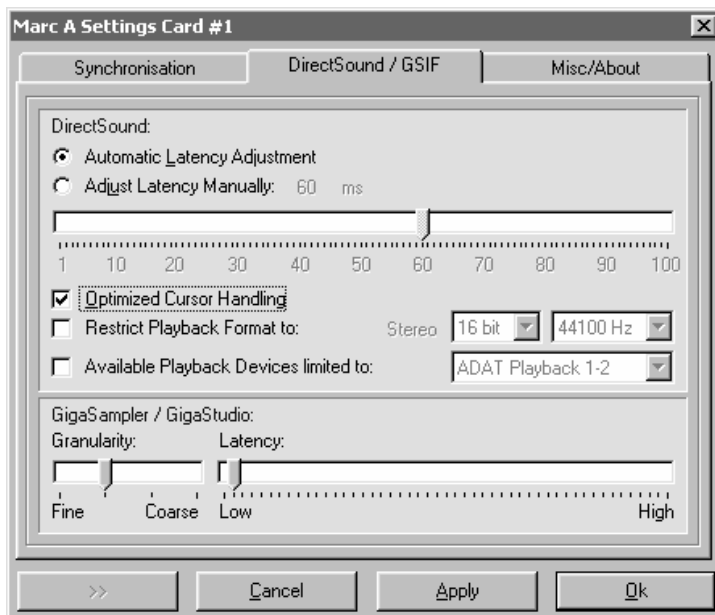### 7.1.2. Audio Settings on Windows™ 98/ME

7.1.2.1. "DirectSound / GSIF" Tab

In this tab you determine the settings for applications which use DirectSound or the Tascam GigaSampler Interface (GSIF) as driver interface. This tab is available on Windows 95/98/ME systems only.

7.1.2.2. "DirectSound" Section

Some information in advance: The ideal settings for the usage of DirectSound depend on some factors of your PC. Because of this we are not able to give you any ideal settings, but only information on what's behind the possibilities of the settings. You will have to find out by testing which settings are the most suitable ones for you. By using the default settings a trouble-free operation can be guaranteed.

"Adjust Latency Manually"

Moving the latency control changes the size of the audio buffer on the card. This way latency is changed during playback of audio signals. If the settings for the latency period is too low, it may result in interruptions. In this case you will have to increase the latency period.

"*Restrict playback format to:*"

When using certain audio programs with DirectSound it may happen, that the DirectX system attempts to open the driver using a different sample format, than the audio programs intends to use. This results in a real time conversion of the signal format by the DirectX system. This conversion may have negative influence on the sound and system performance.
If a playback using DirectSound sounds falsified, you will have to set the signal format, which you use in the audio software. This way the DirectX system is forced to exclusively open the driver using the set format in the MARC A Manager and the real time conversion is inapplicable.

"Available Playback Devices limited to:"

Here you adjust the (stereo) output for DirectSound applications, which have no possibility for automatic selection of the output. The output of the audio signals is then restricted to this (stereo) output.

7.1.2.3. "GigaSampler / GigaStudio" Section

We cannot inform you about the ideal settings for GSIF but will explain to you, what the terms stand for. In order to find the most suitable settings you will have to test.

"Granularity" and "Latency"

The granularity specifies the size of the audio data blocks, which are transferred to the hardware by the GigaSampler. The smaller the latency was set, the smaller will have to be the data blocks, in order to avoid interruption during playback (granularity: "*Fine*"). Though the system is more stressed because of the larger data overhead during the transfer, the reaction time for the sampler playback is shorter.

Larger data blocks (granularity: "*Coarse*") stress the system less, but the reaction time is longer and easier leads to interruptions during playback.

For ideal results try to set the values of the two faders as low as possible.

7.1.2.4. "Others / Info" Tab

"Disable messages of device conflicts"

Here we provide you with the possibility to display or suppress error messages of clock and sample rate problems as well as device problems. By standard the display of error messages is suppressed.

We recommend to enable the display only when you face problems during the start of recording or playback, or your audio software outputs error messages which refer to unknown external errors.

7.1.3. Audio Settings on Windows™ 2000/XP
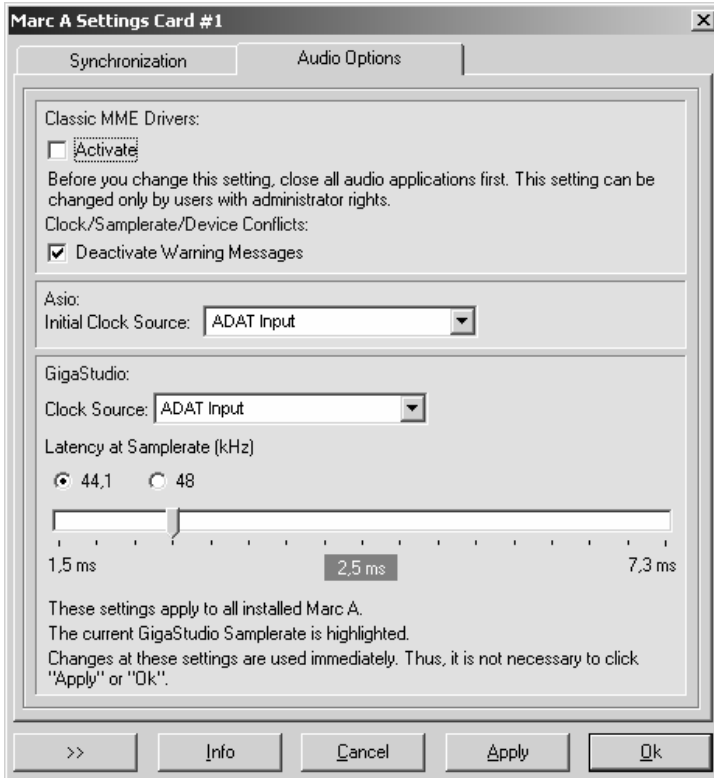
7.1.3.1. WDM, MME and DirectSound

Audio applications, who do not use ASIO, GSIF or direct WDM-Audio (like Cakewalk Sonar), communicate with Microsoft MME or Microsoft DirectSound drivers instead of communicating with the MARIAN drivers directly. The Microsoft drivers in turn communicate with the MARIAN WDM-Audio drivers.

For audio applications, who use these Microsoft drivers, apply:

- They can playback simultaneously using the same playback device. The Microsoft Kernel Mixer mixes the playback signals of the applications into one stereo stream and routes it to one physical output device. If playback takes place with different sample rates, the Microsoft Kernel Mixer converts the sample rate of the playback streams to the highest sample rate required.

> **Hint**: You can improve the quality of this conversion! (See Control Panel | Sounds and Audio Devices | Audio | Sound Playback | Advanced | System Performance)

- The described simultaneous playback of different applications fails, if the output device is already in use by ASIO, GSIF, "Classic MME" or direct WDM-Audio.

- The number of the available recording and playback devices is limited:

  - on Windows 2000 to 10 devices each (20 channels)
  - on Windows XP to 32 devices each (64 channels)

  This limitation applies to the number of all audio devices installed in the system. You can use "Classic MME" or ASIO to walk around this limitation.

- The minimum possible latency is limited to app. 30ms through the Microsoft Kernel Mixer architecture.
  Use ASIO or GSIF to walk around this limitation.

- Some audio applications show additional audio devices with their names appended by "(3+4)", "(5+6)" or "(7+8)". This is caused by an anomaly of the Microsoft MME/DirectSound system. Please ignore these devices and do not use them.

- The Microsoft DirectSound or the Microsoft MME devices do not work with the Driver Settings for Start/Stop Synchronization "Synchronous" and "SyncBus". Activate these settings only if you are sure that you work with ASIO, GSIF or "Classic MME" only! Otherwise, the entire Microsoft WDM-Audio System could freeze. This circumstance cannot be influenced by the driver.

  > **Attention**: The Windows System Sounds are working with DirectSound or MME via WDM-Audio if you do not set a classic MME device as standard playback device!

- Some audio applications require the MARIAN driver because of its additional hardware support functions:

  - Hardware Punch In Monitoring
  - Hardware Pitch Support
  - Hardware Audio Signal Routing
  - Hardware Audio Signal Level Measurement

These functions are not supported by the Microsoft drivers. Please supply this audio applications with the "Classic MME" devices.

7.1.3.2. Section "Classic MME Drivers"
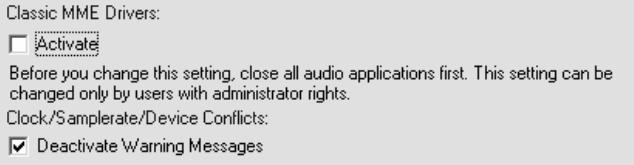
The MARIAN driver enables you to use the former MME driver interface, which was conventional until WDM-Audio was introduced, as an addition to all other driver interfaces. From now, we call this interface "Classic MME". You can activate "Classic MME" within the Marc A Manager settings. If this driver interface is active, you get additional recording and playback devices in the appropriate device lists. These devices have the name suffix "(MME)". The "Classic MME" device have these advantages:

- The number of the recording and playback devices available is NOT limited (see above)
- These devices support the MARIAN hardware support functions (see above)
- These devices work much faster and have better latency values than the MME via WDM-Audio devices
- These devices ensure correct start/stop synchronization. This is not possible with DirectSound or MME via WDM-Audio!

Using the last option in this section you can determine whether or not you want to receive error and warning messages from the Classic MME driver. You should enable these messages only to get some support from the driver when you are on error hunting. Else these messages will occur very often when you start some kind of audio applications.

7.1.3.3. Section "Asio"
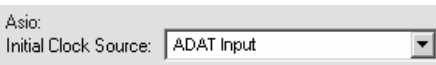
Some ASIO applications do not allow to determine the clock source for an audio setup. In that case you can set the desired clock source here before you start such kind of application. With respect on the multi client capabilities of the Marc A driver it is not possible to change the clock source here for running ASIO applications.

7.1.3.4. Section "GigaSampler/GigaStudio"


```
GigaStudio:

Clock Source: [ADAT Input          ▼]

Latency at Samplerate (kHz)

  ⦿ 44,1    ○ 48

[▮▮▮▮▮▮▮▮▯                          ]
 ·    ·    ·   ·   ·   ·   ·   ·   ·   ·   ·   ·   ·
1,5 ms              [ 2,5 ms ]              7,3 ms

These settings apply to all installed Marc A.
The current GigaStudio Samplerate is highlighted.
Changes at these settings are used immediately. Thus, it is not necessary to click
"Apply" or "Ok".
```
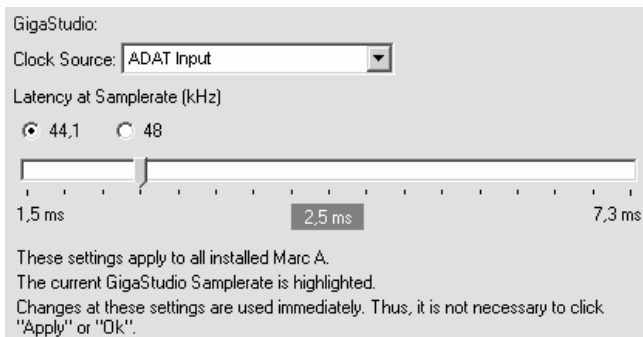
Here you can set all options which are relevant if you use "Tascam GigaStudio". All these settings can you change even if "GigaStudio" is working with the Marc A. This a great chance to validate the settings acoustically.

The first options determines which clock source "GigaStudio" is using for audio recording and playback.

In section "Latency at Samplerate (kHz)" you can setup the latency between a Tascam GigaStudio MIDI event and the resulting sound playback.

Move the latency slider right, to increase the latency value. Mode the latency slider left, to decrease the latency value.

You should increase the latency value only, if you hear drop outs during the GigaStudio playback. As a result of increasing the latency value, the GigaStudio playback becomes more stable.

The latency also depends on the playback sample rate. You can choose another sample rate to **show** the resulting latency value – you will not change the sample rate. The actual GigaStudio sample rate is shown in blue color.

8.   ASIO Settings

The picture on the right shows the ASIO controls panel of the MARC A driver. You can open this panel within your ASIO application program. Please refer the manual of this program to learn where the ASIO control panel can be opened.

By enabling the checkboxes in front of the entries of the inputs and outputs you determine, which devices the ASIO application can 'see' and use.
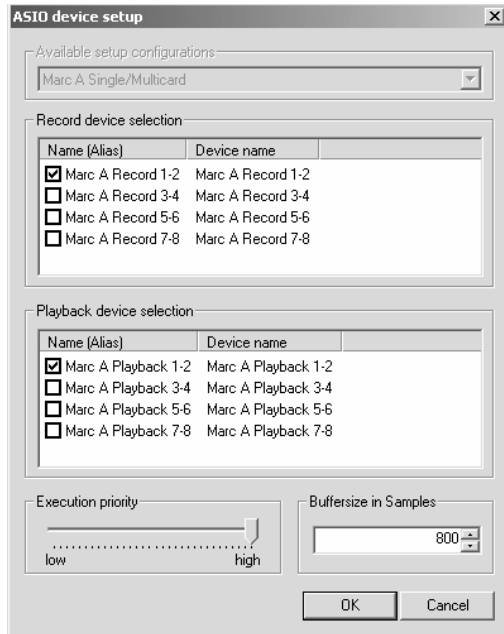
You can change the listed "*Name (Alias)"* in the first column by double-clicking the entry. This way you can rename it as you like.

Hint: Only activate the playback and recording devices which you really want to use within you project! Each activated device reserves system resources and CPU load – even if you do not use the device in you audio project.

Should you use an ASIO compatible audio software, you may have to perform additional settings in the ASIO configuration of the software. For details please refer to the software manual.

When you enable a Marc A recording device, please ensure that there is a valid signal at the ADAT input. If this is not available, neither the recording nor the playback of any device will work.

The slider "*Execution priority"* determines in "*high"* position, that the transmission of ASIO data is given a high processor priority. In the position "*low"* it is given to real time calculation of Plug-Ins.

Every audio application uses buffers to transfer the audio data to and from the hardware. These buffers work like containers. During playback the application fills the container with the playback data and the driver empties this container. The size of the container determines the latency

between a live event (like a midi event or live recording) and the resulting playback. The latency grows with the size of the buffer.

You can setup the buffer size with the field *"Buffersize in Samples"*. Here you can tell the driver how many samples should fit in the container. The resulting latency value, expressed as a time value like milliseconds, depends on the sample rate. The most ASIO audio applications show this value when you close the ASIO setup dialog of the driver.

9.   Software Samplers and Synthesizers

Software samplers and synthesizers normally supply MIDI output devices for other audio applications which want to use the software sampler as an instrument. If such an application starts, then normally this application opens the MIDI output devices and this in turn causes the software sampler to initialize its audio engine with the configured audio outputs.

This occurs even if you have NOT started the software sampler application.

In this situation the software sampler and the other audio application may conflict, if they use the same audio output. If you use other driver interfaces than Microsoft MME or DirectSound, you can use the Marc A with different audio applications at the same time but never the same output device simultaneously.

Therefore this hint:

- First start the software sampler/synthesizer and setup an audio output device which will NOT be used by the other audio application. Start the audio application (sequencer) afterwards.

- If you start an ASIO or GSIF application the first time, then it will use the device "Marc A 1-2" for recording and playback.

The Windows Multimedia System also supplies a software synthesizer called "Microsoft GS Wavetable SW Synth". This synthesizer has also a MIDI output port with the same name. Since there is no explicit setup for the output device, the "Microsoft GS Wavetable SW Synth" always uses the standard playback device configured in "Control Panel | Sounds and Multimedia | Audio". In the device setup you can see the window shown on the right.

10.   Technical Data

- Short PCI-card; 32-bit 33MHz, 5V
- 1 ADAT™ input (optical via TOSLINK)
- 1 ADAT™ output (optical via TOSLINK)
- Sample formats: 8, 16, 20, 24 Bit mono/stereo
- Sample rates for recording: 40 - 51 kHz
- Sample rates for playback: 22 – 58 kHz

11.   Service and Support

If you have any questions or problems during the installation or operation of your MARC A please perform the following steps:

1.   Ensure that you always have the latest driver installed. You find the current driver files on the following internet page: **www.marian.de/en/downloads**

2.   If any questions still remain, you can contact us via the internet using our support form at: **www.marian.de/en/support**

or give us a phone call: **+49 341 5893222**.

Interesting news, information and driver updates you can find here: **www.marian.de**.