



User's Guide



The MARC X conforms the following standards:
EN 55022: 1998 + A1: 2000 + A2: 2003; class A
EN 55024: 1998 + A1: 2000 + A2: 2003; class A

In order for an installation of this product maintain compliance with the limits of a class A device, shielded audio cables must be used, not longer than 50 cm. Attention: This is a device of the class A and can cause interference to radio or television reception within the residential area. The user is encouraged to try to correct the interference by suitable measures.

© March 2007, v2.2 MARIAN.
Hardware Design by MARIAN
"SyncBus" developed by MARIAN

All rights reserved. No part of this User's Guide may be reproduced or transmitted in any form or by any means, electronically or mechanical, including photocopy, translation, recording, or any information storage and retrieval system, without permission in writing from MARIAN.
All trademarks are the property of the respective owners.

MARIAN is not liable for any damage to the software, hardware and data and costs resulting from it, which are caused by improper handling or installation of the hardware.

Technical changes are reserved.

Table of Contents

1.	Features	3
2.	Scope of Supply	4
3.	System Requirements	4
4.	Ports	5
4.1.	MARC X PCI Card	5
4.1.1.	Audio Ports	5
4.1.2.	SyncBus Port	6
4.1.3.	Ports for Extender Cards	6
4.2.	MIDI/Wordclock Extender Card "MWX"	7
4.3.	Analog I/O Extender Card "ANX"	8
4.4.	AES/EBU Extender Card "DGX"	8
5.	Hardware Installation	8
6.	Driver Installation	10
6.1.	About MME, DirectX, WDM-Audio, ASIO and GSIF	10
6.2.	Installation on Windows 95 (Release 950 and 95a)	11
6.3.	Installation on Windows 95 (Release 95b and 95c)	11
6.4.	Installation on Windows 98	12
6.5.	Installation on Windows ME	12
6.6.	Installation on Windows 2000	13
6.7.	Installation on Windows XP	13
7.	MARC X Manager	14
7.1.	Audio Settings	14
7.1.1.	Operating the Control Elements	15
7.1.2.	Mixer	16
7.1.2.1.	Input and Playback Channels	17
7.1.2.2.	Master Section	18
7.1.2.3.	Configuring the View	19
7.1.3.	Controlling the Outputs	19
7.1.3.1.	Output Channels	20
7.1.3.2.	Optical Output	21
7.1.3.3.	Configuring the View	21

7.2.	Operating Mode, Clock, MIDI and other Settings	22
7.2.1.	Operating Modes	22
7.2.2.	Clock Settings	23
7.2.3.	Internal Clock	25
7.2.4.	Wordclock Output	25
7.2.5.	Start/Stop Synchronization	26
7.2.6.	Clock and Sample Rate Conflicts	27
7.2.7.	Setting up the driver interfaces in Windows™ 95/98/ME	28
7.2.7.1.	Section " <i>Direct Sound</i> "	28
7.2.7.2.	Section " <i>GigaSampler/GigaStudio</i> "	29
7.2.8.	Setting up the driver interfaces in Windows™ 2000/XP	30
7.2.8.1.	MME and DirectSound	30
7.2.8.2.	Section " <i>Classic MME Drivers</i> "	31
7.2.8.3.	Section " <i>GigaSampler/GigaStudio</i> "	31
7.2.9.	MIDI Settings	32
7.2.9.1.	Section " <i>Command Filters</i> "	33
7.2.9.2.	Section " <i>System Data Filters</i> "	34
8.	ASIO Settings	35
9.	Software Samplers and Synthesizers	37
10.	Technical Data	38
11.	Service and Support	39

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 X easier for you.

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

1. Features

Your MARC X provides special characteristics. In the following you will get a short overview:

- Stereo analog I/O, 24 bit/96 kHz
- Stereo digital I/O S/PDIF, 24 bit/96 kHz
- 8-channel digital I/O ADAT, 24 bit/48 kHz
- Port for an internal digital audio CD/DVD drive
- Port for an internal analog audio CD/DVD drive
- 2 MIDI I/Os
- Wordclock I/O, Superclock I/O
- 32-channel DSP mixer, hardware-based, latency-free
- Latency-free and flexible monitoring
- DAT marker support
- MARIAN SyncBus for synchronization with other MARIAN sound systems
- Multiple card support (up to 4 cards can be installed)

The analog I/O extender card "ANX" (not included in the scope of supply – separately available as accessory) further provides the following characteristic:

- 2 additional stereo analog I/Os, 24 bit/96 kHz

The Digital I/O extender card "DGX" (not included in the scope of supply – separately available as accessory) provides the following characteristic:

- 2 AES/EBU Inputs and Outputs (4 channels)
- 2 Input Sample Rate Converter

2. Scope of Supply

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

- MARC X PCI card
- Extender card for MIDI and Wordclock "MWX"
- Ribbon cable for internal connection of "MWX" with the MARC X
- Cable for connecting the digital audio signal of an internal digital audio CD/DVD drive
- MIDI cable adapter
- User's Guide
- CD-ROM incl. drivers

If you have additionally purchased the analog I/O extender card "ANX", the following components are supplied:

- Analog I/O extender card "ANX"
- Ribbon cable for the internal connection of the "ANX" and MARC X
- Power cable for connecting the "ANX" to the internal power supply of the PC

If you have additionally purchased the AES/EBU I/O extender card "DGX", the following components are supplied:

- DGX extender
- XLR breakout cable with 2 AES/EBU connectors
- Ribbon cable for the internal connection of the "DGX" and MARC X
- Power connector
- Driver CD
- DGX Manual

3. System Requirements

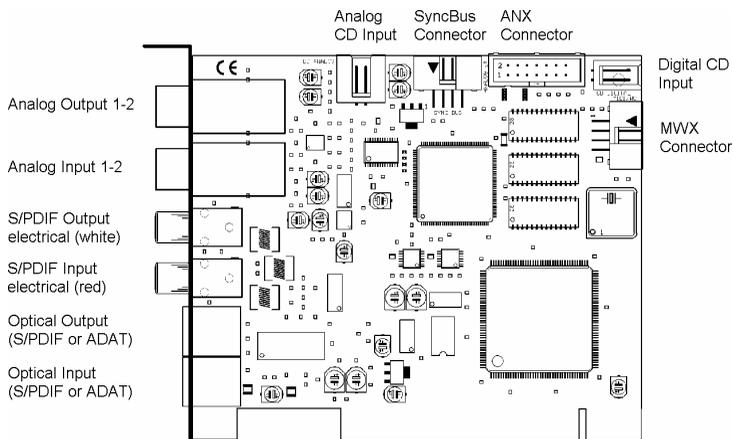
In order to be able to operate the MARC X, your PC should meet the following requirements:

- AT compatible PC with a spare PCI slot
- Pentium or AMD processor with 350 MHz or higher
- Windows™ 95/98/ME/2000/XP
- Microsoft DirectX 9
- For extender cards "MWX", "ANX" and "DGX":
 - a spare slot in the PC housing – PCI or ISA slot is not required
- For extender card "DGX": Windows™ 2000/XP

Please note, that the audio software you use may ask for different requirements.

4. Ports

4.1. MARC X PCI Card



The ports of the MARC X.

4.1.1. Audio Ports

Analog devices are connected to the MARC X using 1/4" TRS jacks. Please exclusively use high-quality and shielded cables in order to achieve maximum audio quality. You can connect additional analog devices using the optional analog I/O extender card "ANX".

For connecting S/PDIF devices, you can use electrical (RCA) as well as optical jacks (TOSLINK).

ADAT devices are connected using optical ports (TOSLINK).

If you want to use optical digital ports, you need optical connection cables, which you can purchase at specialist shops.

For the processing of the audio signal of your CD/DVD-ROM drive, you can either use an analog input or an electric digital input. Basically, you should prefer using a digital CD input, because this kind of connection offers optimum audio quality. For information of which ports are provided by your CD/DVD-ROM drive, please refer to the appropriate manual.

4.1.2. SyncBus Port

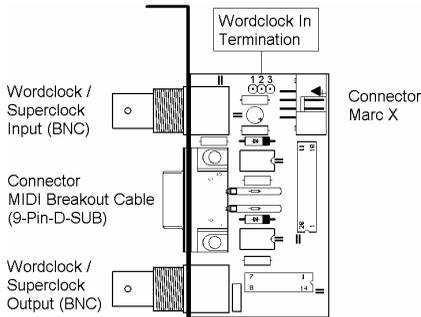
If you use several MARC cards or other cards developed by MARIAN, you can connect the SyncBus ports of the individual cards using a cable. This SyncBus cable is available as accessory. If you want to order one, please get in touch with our Sales Department (see attachment).

The SyncBus of your MARC X has two tasks: on one side it synchronizes the sample rate and ensures that several cards run sample-accurately and synchronously when they are operated concurrently. On the other side it takes care of the synchronous starting and stopping of several internal audio devices. For details on the settings see chapters 7.2.2 and 7.2.5.

4.1.3. Ports for Extender Cards

The MARC X further provides a port for the MIDI/Wordclock extender card "MWX" and a port for the optional analog I/O extender card "ANX" and the optional AES/EBU extender card "DGX". "ANX" and "DGX" can not be operated simultaneously.

4.2. MIDI/Wordclock Extender Card "MWX"



The picture on the left shows all ports of the "MWX".

Using the Wordclock input you can synchronize the MARC X in a Wordclock environment. If the extender is connected at the end of the Wordclock bus, then the Wordclock input of the extender needs to be terminated. This is assured by

setting the jumpers on the pins 2-3 of the jumper block. If the Wordclock termination is not required, then please set the jumpers on the pins 1-2 of the jumper block.

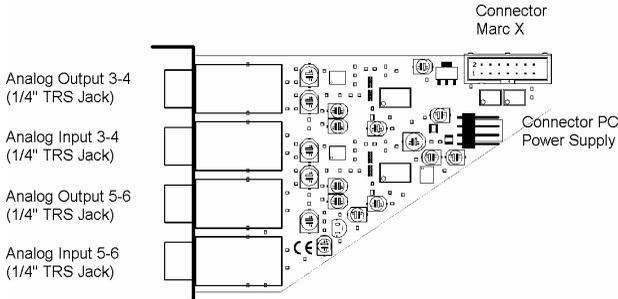
The Wordclock input also accepts Superclock. An according configuration is required and is performed in the clock settings of the driver software (see chapter "7.2.2 Clock Settings").

For using the Wordclock input with Superclock, a termination is urgently required to ensure a stable operation!

The Wordclock output allows the operation of the MARC X as clock master in a Wordclock environment. It can also be configured as Superclock output (see chapter "7.2.4 Wordclock Output").

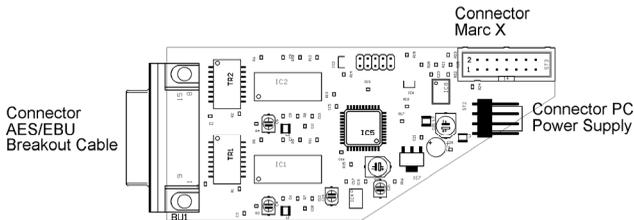
You can connect the supplied MIDI cable to the 9-pin D-SUB port. It has four DIN jacks for two MIDI I/Os. Connect your external MIDI device to these jacks.

4.3. Analog I/O Extender Card "ANX"



This extender card is not included in the MARC X scope of supply. You can order this card separately and you will receive additionally four analog I/Os in 24 bit/96 kHz.

4.4. AES/EBU Extender Card "DGX"



This extender card is not included in the MARC X scope of supply. You can order this card separately and you will receive additionally two AES/EBU I/O and Input Sample Rate Converters.

5. Hardware Installation

Following the steps described below, you can install the MARC X, the MIDI/Wordclock extender card "MWX" and the optional analog I/O extender card "ANX" on the same PC. Please pay attention to the pictures and information about the ports in chapter "4 Ports".

1. Turn off your PC and all devices connected to it, disconnect the power supply.
2. Open the case.
3. To insert the MARC X:
 - Remove the MARC X 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.

- Insert the card carefully and rectangular into a spare PCI slot. Ensure that the card was inserted properly into the slot.
 - Screw the card on the slot bracket to the case.
 - If you want to operate the MARC X concurrently with other MARIAN cards, then all cards need to be connected by the *SyncBus*. For this, plug the port of a SyncBus cable to the according port of the MARC X, and the others on the according ports of the other cards.
4. To insert the MIDI/Wordclock extender card "*MWX*":
 - Remove the extender card 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.
 - If you want to use the Wordclock input, then set the Wordclock termination according to your requirements.
 - Plug the ribbon cable into the matching connector.
 - Insert the extender card into a spare case's output, which can also belong to an unused slot on the motherboard. Ensure that the extender was inserted properly into the slot.
 - Screw the extender card on the slot bracket to the case.
 - Connect the other end of the ribbon cable with the MARC X.
 5. To insert the optional extender cards "*ANX*" or "*DGX*":
 - Remove the extender card 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.
 - Plug the power cable into the designated port.
 - Plug the "*ANX*"/"*DGX*" ribbon cable into the designated port.
 - Screw the card on the slot bracket to the case.
 - Connect the other end of the power cable to the spare port inside your PC.
 - Plug the other end of the ribbon cable into the MARC X
 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 X with many audio software applications. The Marc X driver software with all its interfaces makes that possible. An audio application uses a driver interface to transfer the audio data and the MIDI data to and from the Marc X 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 “7.2.7 Setting up the driver interfaces in Windows™ 95/98/ME” and “7.2.8 Setting up the driver interfaces in 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 the hardware.
2. At startup Windows automatically recognizes 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 `\marcx\win9x\english` directory and confirm your selection. Windows copies the driver files and installs MARC X.
7. After finishing the installation, your MARC X 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 the hardware.
2. At startup Windows automatically recognizes 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 to the `\marcx\win9x\english` directory on the CD and confirm it by clicking the “*OK*” button.
6. Windows copies the driver files and installs the MARC X. 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 X is ready to be operated without having to restart the system.

6.4. Installation on Windows 98

1. Start your PC after installing the hardware.
At startup Windows automatically recognizes the newly installed device and starts the Hardware Wizard.
2. During the start of Windows the Hardware Wizard searches for new drivers. Click *"Next"* to start the search.
3. Enable the *"Search for the best driver for your device"* option and confirm by clicking *"Next"*.
4. Insert the supplied CD in your CD-ROM drive.
5. Select your CD-ROM drive in the next dialog, change into the `\marcx\win9x\english` directory and click *"Next"*. The Hardware Wizard searches the CD-ROM for the appropriate drivers and finally displays the names of the drivers.
6. Click *"Next"*; Windows copies the driver files and installs the MARC X.
7. After finishing the installation click *"Finish"*. The driver will be enabled and the MARC X is ready to be operated without having to restart the system.

6.5. Installation on 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 X drivers on the CD. Choose the drivers for Windows 95/98/ME (folder `"\marcx\win9x"`) and your preferred language. Confirm this dialog with *"Next"*.
6. Windows copies the driver files and installs the MARC X.
7. After finishing the installation click *"Finish"*. The driver will be enabled and the MARC X is ready to be operated without having to restart the system.

6.6. Installation on Windows 2000

1. Start your PC after installing the hardware.
2. Logon using administrator rights.
3. At startup Windows automatically recognizes 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. Enable the *"Search for the best driver for your device"* option and confirm by clicking *"Next"*.
6. Insert the supplied CD in your CD-ROM drive.
7. Confirm the next dialog with *„Next“*
8. Confirm the next window also with *„Next“*
9. Ignore the message *„Digital signature not found“* and continue the installation with *„Yes“*
10. Windows copies the driver files and installs the MARC X.
11. After finishing the installation click *"Finish"*. The driver will be enabled and the MARC X is ready to be operated without having to restart the system.

6.7. Installation on Windows XP

1. Start your PC after installing the hardware.
2. Logon 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 X.
8. After finishing the installation the MARC X is ready to be operated without having to restart the system.

7. MARC X Manager

The MARC X Manager enables you to perform audio, MIDI as well as clock and synchronization settings.

After successful driver installation the symbol shown on the right appears for the MARC X Manager in the Windows taskbar. With a simple mouse-click you can open a menu, in which you can select what kind of settings you want to change.



„Mixer“ opens the 32-channel DSP mixer, in which all available audio signals are mixed to three sums. In chapter "7.1.2 Mixer" these settings are explained in more detail.



„Output Control“ opens the output control window, in which you can determine the signal routing for outputs as well as the output level (see chapter "7.1.3 “).

„Settings“ opens a dialog box, in which you can determine the operating mode, and also change the settings for clock and synchronization, DirectSound and GSIF, as well as MIDI. In chapter "7.2 Operating Mode, Clock, MIDI and other Settings“ these settings are explained in more detail.

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

„Exit“ ends the execution of the MARC X Manager software.

The user manual “DGX Extender” contains more information about the integration of “DGX” in the Marc X Manager.

7.1. Audio Settings

The MARC X Manager provides the "Mixer" and "Output Control" windows, in which you can perform all audio settings. The "Mixer" window visualizes the DSP based 32-channel mixer of the MARC X. This mixer is integrated into the card's hardware and allows instantaneous mixing of all available input signals as well as of all playback signals to three stereo sums without additional CPU load. The "Output Control" window visualizes all available physical outputs. It allows to route any output to one of the available input signals, playback signals as well as one of the mixer's sums and to determine the volume of an output. Both windows are subdivided in stereo channels.

7.1.1. Operating the Control Elements

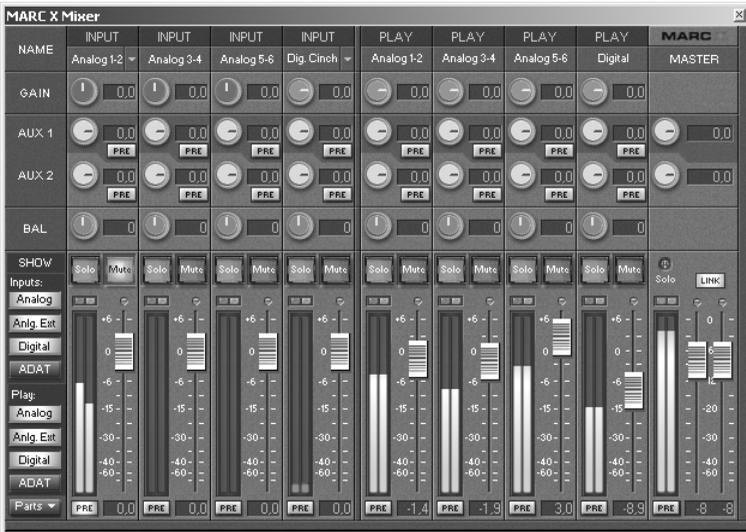
The channels contain control and display elements.
In the following you will find some operating instructions:

- All rotary-buttons, faders and switches can be operated by using the left mouse button.
- Active switches are illuminated.
- Rotary-buttons and faders can be reset to the standard value, e.g. 0 dB, by double-clicking the left mouse button.
- Each rotary-button and fader has a numeric value display, which can be edited by double-clicking the left mouse-button followed by your input using the keyboard. The input is finalized by the pressing the *<Enter>* key or by selecting another control element.
- Each channel contains a stereo level indicator with peak LED's (top) and a *"PRE"* switch (bottom). If the switch is active, the level of the source signal is indicated.

The peak LED's are switched permanently to *"PRE"* and signal a clipping of the source signal.

If level or peak indicators are switched to *"PRE"*, the level of the signal prior the GAIN control is measured. The only exception are the analog input channels. Since the GAIN is controlled in the AD converter directly, *"PRE"* switched level or peak indicators measure the level of the signal after the GAIN control but prior the fader.

7.1.2. Mixer



The "MIXER" allows mixing of available input signals and playback signals to three stereo sums – to AUX1, AUX2 and the master sum. The Mixer is divided in an input, a playback and a master section.

Within the input section an "INPUT" channel is assigned to each of the Marc X inputs.

Within the playback section a "PLAY" channel is assigned to each of the identical names playback devices, which can be found in the audio device lists of your audio application.

Within each channel you can set the percentage of the channels signal at the sum signals "AUX 1", "AUX 2" and the main sum. The master section of the mixer contains a sum channel "MASTER". Here you can set up the volume of the three sum signals.

The number of displayed channels depends on the operating mode of the MARC X (see chapter "7.2.1 Operating Modes"). In the operating mode "NON-ADAT Mode" ADAT inputs and outputs are not available and will not be displayed in the mixer.

The input channels "Analog 3-4" and "Analog 5-6" or "AesEbu 1-2" and "AesEbu 3-4" are displayed only, if the according extender card is connected.

7.1.2.1. Input and Playback Channels

The input section of the mixer contains a separate "INPUT" channel for each available input signal. The playback section of the mixer contains a separate "PLAY" channel for each available playback device. In the upper part of each channel, the channel is labeled with its name. Thus, the corresponding input or playback channel can be uniquely identified. If an input has several connection options (e.g. digital input: optical and electrical, digital CD), then these options are offered at the top in the channel in a dropdown list. Each channel provides the following options:

- Using the "GAIN" control you can set the signal's amplification prior the fader and the AUX controls.

The "GAIN" controls of the analog input channels are in a darker red color in contrast to all other channels. These "GAIN" controls set the pre-amplification directly in the analog-digital converter. This changes the level for all applications globally, which take recordings from this input. The "GAIN" controls of the ADAT, the digital and the analog play channels influences the signal level only within the mixer.

- Using the yellow rotary-buttons "AUX 1" and "AUX 2" the percentage of the signal in the according sum can be determined. A "PRE" switch is assigned to each of the AUX controls. If this switch is active, the fader has no influence on the percentage of the signal in the according sum.
- Using the blue rotary-button "BAL" (balance) you can regulate the volume ratio between the left and the right channel of the stereo signal.
- Using the fader in the lower part of the channel you can set the signal's percentage on the main sum signal.
- Using the "Mute" switch you can mute the signal of the channel for all sum signals. This way the signal of the channel is no longer contained in the AUX sums or the main sum.
- If the "SOLO" switch is active in one or more channels, the main sum contains exclusively the signals of these channels. The "SOLO" switch has no influence to the AUX sums.

Each channel contains a stereo level indicator with peak LED's (top) and a "PRE" switch (bottom). If the switch is active, then the level of the source signal is indicated.

There are some situations where the signal level cannot be displayed for digital input channels. The cause is an improper clock situation. In this case, two red LED's illuminate the bottom of the level indicator. Please refer chapter "7.2.2 Clock Settings" to learn more about it.



The **GAIN** controllers of the analog inputs influence the level of the source signal.

The peak LED's are switched permanently to **"PRE"** and signal a clipping of the source signal.

If level or peak indicators are switched to **"PRE"**, the level of the signal prior the **GAIN** control is measured. The only exception are the analog input channels. Since the **GAIN** is controlled in the AD converter directly, **"PRE"** switched level or peak indicators measure the level of the signal after the **GAIN** control but prior the fader.

7.1.2.2. Master Section

The **MASTER** section is on the right side of the mixer. It contains the sum channel **"MASTER"**. In this channel the volumes of the sum signals can be regulated:

- Using the two yellow rotary-buttons you can regulate the volume of the sum signals **"AUX 1"** and **"AUX 2"**.
- For setting the volume of the main sum you can use the two faders in the bottom part of the channel. They are assigned each to the left and right channel of the stereo sum.
- Using the **"Link"** switch the two faders are connected with each other and one fader controls the other. In this case relative differences in volume are maintained.
- If at least one channel has activated the **"Solo"** switch, the **"Solo"** LED in the master section illuminates.
- The channel contains a stereo level indicator with peak LED's (top) and **"PRE"** switch (bottom). If this switch is active, then the level of the main sum signal prior the fader is indicated.



The peak indicator is switched permanently to **"PRE"** and signals a clipping of the main sum signal prior the fader.

7.1.2.3. Configuring the View

Using the "SHOW" panel you can adjust the appearance of the mixer. You can find it on the bottom left of the mixer. It contains switches, which allow to display and hide individual channel groups or channel elements. For the "INPUT" and "PLAY" channels each channel group can be displayed or hidden separately. The following channel groups are distinguished:

- "Analog" contains the channel "Analog 1-2"
- "Anlg Ext" contains the channels "Analog 3-4" and "Analog 5-6" of the optional analog I/O extender card "ANX"
- "Digital" contains the channel "Digital"
- "ADAT" contains the channels "ADAT 1-2" to "ADAT 7-8"



Using the switch "Parts" you can activate and deactivate the display of control elements for all channels of the mixer.

7.1.3. Controlling the Outputs



All output settings are performed using the "OUTPUT CONTROL" window. It contains a channel each for all available physical outputs. Within the "OUTPUT CONTROL" window all input signals, playback signal and sum signals can be assigned absolutely flexibly to the available outputs.

The number of displayed channels depends on the operating mode of the MARC X (see chapter "7.2.1 Operating

Modes"). In the "NON-ADAT Mode" no channels are displayed which are related to ADAT channels.

The channels "Analog 3-4" and "Analog 5-6" or "AesEbu 1-2" and "AesEbu 3-4" are displayed only, if the according extender card is connected.

7.1.3.1. Output Channels

For each available physical output of the MARC X, the "OUTPUT CONTROL" window contains an "OUTPUT" channel, where the source signal can be selected and the volume for this output can be set:

For selecting the source signal ("SOURCE"), which is played back via this output, the channel in the upper part contains the switches "Play", "Master", "Aux 1", "Aux 2" and "Input". For each channel only one of these switches can be activated.

If you activate "Play", the playback signal of an audio application is routed to the output. Using the drop down list below, you can determine whose playback signal can be heard. Each list entry corresponds to a Marc X playback device name, which is also used by your audio application. Thus, e.g. you can hear the playback signal of "Marc X Digital" at the analog output 1-2.

The central switches provide the selection of one of the mixer's sum signals.

If you activate "Input", the signal of an Marc X input is routed to the output. You can use the drop down list below to specify the required input.

- For setting the volume you can use the two faders in the bottom part of the channel. They are assigned each to the left and right channel of the output signal.
- Using the "Link" switch the two faders are connected with each other and one fader controls the other. Relative differences in volume are maintained.
- Using the "Mute" switch you can mute the output signal.
- Each channel contains a stereo level indicator with peak LED's (top) and "PRE" switch (bottom). If the switch is active, the level of the source signal is indicated.

The peak indicator is switched permanently to "PRE" and signals a clipping of the source signal.

If level or peak indicators are switched to "PRE", the level of the signal prior the fader is measured.



7.1.3.2. Optical Output



In the "*Opt. OUT*" panel you can select which source signal is played on the optical output of the MARC X. You can find the panel below the "*SHOW*" panel. It contains two switches, which can be used for determining whether "*ADAT*" or "*S/PDIF*" is played. Only one of the switches at a time can be activated.

7.1.3.3. Configuring the View

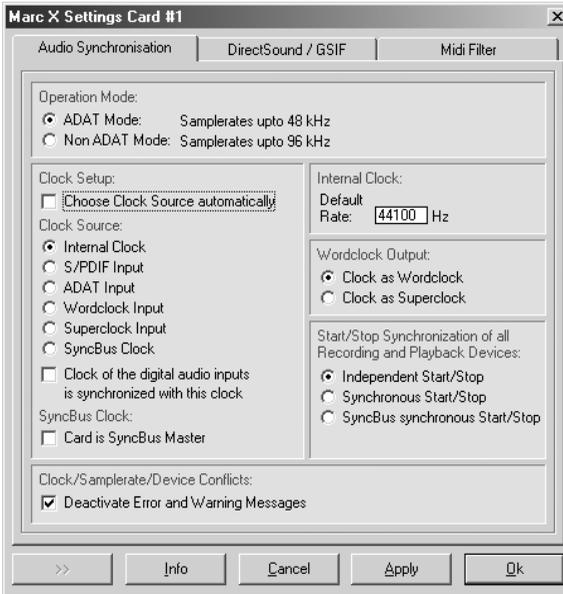


Using the "*SHOW*" panel you can adjust the appearance of "*OUTPUT CONTROL*". You can find it on the bottom left of the window. It contains switches, which allow to display and hide individual channel groups.

The following channel groups are distinguished:

- "*Analog*" contains the channel "*Analog 1-2*"
- "*Anlg Ext*" contains the channels "*Analog 3-4*" and "*Analog 5-6*" of the optional analog I/O extender card "*ANX*"
- "*Digital*" contains the channel "*Digital*"
- "*ADAT*" contains the channels "*ADAT 1-2*" to "*ADAT 7-8*"

7.2. Operating Mode, Clock, MIDI and other Settings



You can find the dialog box for these settings by selecting the menu of the MARC X Manager – “Settings”. Here you can determine the operating mode of the MARC X as well as the settings for clock and synchronization, DirectSound and GSIF, and MIDI.

Please note that some settings and labeling are different in Windows™ 95/98/ME and Windows™ 2000/XP.

The user manual “DGX Extender” contains more information about the integration of “DGX” in the Marc X Settings.

7.2.1. Operating Modes

The setting “*Operating Mode*” has global influence on the behavior of the MARC X. The MARC X can be operated in two different modes:

- in “*ADAT Mode*”, for using all inputs and outputs including the ADAT channels. In this mode the sample frequency for all available channels is limited to a maximum of 48 kHz + 15% pitch.
- in “*NON-ADAT Mode*”. This mode excludes the usage of ADAT inputs and outputs. In this mode a sample frequency of up to 96 kHz + 15% pitch is allowed for all remaining channels.

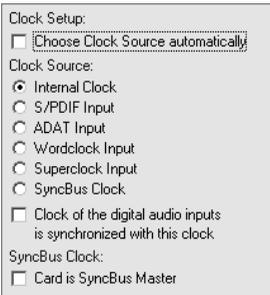
The “ADAT Mode” limits the maximum available sample frequency to 48 kHz + 15% Pitch. The “NON-ADAT Mode” excludes the usage of ADAT channels.

7.2.2. Clock Settings

For digital audio processing a clock is always required. You could easily compare this with the engine of a car. If the engine does not run, you cannot drive the car.

The clock is the engine that powers the digital audio processing. The sample rate is derived from the speed of the clock.

In the "*Clock Setup*" you determine, which of the clock sources is used for recording, playback, mixer and monitoring of the MARC X. The selected clock source generates the sample rate, which is used to perform these functions.



If you have selected "*Internal Clock*", then the MARC X generates the clock itself and is independent from other external devices. This setting is also called *Clock Master*. External devices can receive the clock, which is generated by the MARC X, if they are connected by the Wordclock output, S/PDIF output or ADAT output. These devices can be configured as *Clock Slave* and then work synchronously with the MARC X.

For "*S/PDIF Input*" and "*ADAT Input*" the clock results from the digital audio input signal.

If you select "*Wordclock Input*", the clock on the Wordclock of the "*MWX*" input is interpreted and used as Wordclock.

If you select "*Superclock Input*", the clock on the Wordclock input of the "*MWX*" is interpreted and used as Superclock.

If you select "*SyncBus Clock*", the clock of another MARIAN card, which is configured as clock master, is used.

Please note, that only if you select "*Internal Clock*", the application program is able to determine the sample rate. For all other clock sources the sample rate is determined by external connected devices. In such cases MARC X works as clock slave and consequently synchronously with the external clock.

If you select another clock source as "Internal Clock", please ensure, that a suitable is connected to the according port, switched on and outputs a valid signal or valid clock.

For processing digital audio input signals, e.g. for recording, monitoring, mixing or gathering the input level, it is necessary, that the MARC X locks onto the digital input signal. In order to achieve this, you will have to switch the clock source of the MARC X to the appropriate input, e.g. for using the S/PDIF input, you have to switch to "*S/PDIF Input*" and for using the ADAT input, choose "*ADAT Input*".

Or you activate the option "*Choose Clock Source automatically*". If this option is active, the driver of the MARC X first of all chooses "*Internal Clock*". If you start using one of the digital audio input signals, then the driver of the MARC X will switch to a suitable clock source. The driver switches to "*Internal Clock*" again, when the digital audio input is no longer used.

If the digital signal of the input is based on the used clock source, then the required switch-over to the clock source of the digital audio input is not necessary. Examples:

1. You work with an external ADAT device, e.g. the ADAT converter MARIAN ADCON or the ALESIS ADAT XT. If the "*Internal Clock*" is selected as clock source for the MARC X, it works as clock master. If you configure the MARIAN ADCON as clock slave (AD/DA Mode) or set the clock select of the ADAT XT for instance to "EXT 44.1 kHz", these devices return the receiving clock from the MARC X using its digital audio signal back to the MARC X - the input signal of the MARC X is also based on its own internal clock.
2. You work with ADAT and S/PDIF devices, which have a Wordclock input. Select "*Internal Clock*" as clock of the MARC X (clock master) and connect the Wordclock output of the MARC X (port on "MWX") with the Wordclock inputs of your devices. Configure them, so that they operate as clock slave of the Wordclock input. The clock of the digital audio signals of these devices is now based on the internal clock of the MARC X. Now the MARC X is able to process these signals at its digital inputs. Also a simultaneous processing of both signals is possible, because the clock of these signals is synchronized.
3. You work with external ADAT and S/PDIF equipment, which is embedded in a Wordclock environment with an existing Wordclock master. Select "*Wordclock Input*" for the MARC X, which now operates as clock slave. The digital audio input signals of the MARC X (S/PDIF and ADAT) are now based on the same clock, which you have set to be used ("*Wordclock Input*"). This way you can use S/PDIF and ADAT input signals without having to switch to the according clock source.

You setup this condition in the MARC X driver by using the option "*Clock of digital audio inputs is synchronized with this clock*". Activate this option, if one of the above-mentioned clock configurations can be compared with your conditions. In this case the driver allows to use a digital audio input, even if the suitable clock source is not set.

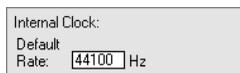
Activate the option "*Card is SyncBus Master*", if the MARC X should operate as clock master for the SyncBus. The clock of the configured clock source will be put on the SyncBus and can be used by other connected MARIAN cards as clock source. These cards then operate as clock slave of the MARC X and consequently synchronously.

If several cards are to be synchronized using the SyncBus, exactly one card has to be configured as master. All other cards have to be configured as slave.

If you work with an ASIO application program, all clock settings will be performed within this program. In this case clock settings performed in the MARC X Manager are ignored. Please refer to the user's guide of your ASIO application program in order to learn how clock sources are selected. In all cases the ASIO driver of the MARC X assumes, that the clocks of the digital input signals are based on the clock source, which you have selected. Internally it activates the option *"Clock of digital audio inputs is synchronized with this clock"*.

7.2.3. Internal Clock

"Default Rate"

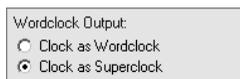


Internal Clock:
Default
Rate: Hz

This input field contains the sample rate, which the internal clock is operated with, as long as the MARC X is not used by recording or playback. If the internal clock is the active clock source, then the mixer as well as the signal routing operates with this sample rate.

Some application programs finish to work with MARC X on a short-term basis, if you stop the recording or playback, or if you re-position the playback cursor (Song Position Pointer) during playback. If you use another sample rate than the configured one, when working with such a program, then setting the *"Default Rate"* could have cracks as a consequence. We recommend to enter the same sample rate in to this input field, which you use for your application program.

7.2.4. Wordclock Output



Wordclock Output:
 Clock as Wordclock
 Clock as Superclock

The Wordclock output of the MARC X always outputs the clock of the currently selected clock source. At this point you determine, whether the clock output of the *"MWX"* should output this clock as Wordclock or as Superclock.

7.2.5. Start/Stop Synchronization

Start/Stop Synchronization of all Recording and Playback Devices:

- Independent Start/Stop
- Synchronous Start/Stop
- SyncBus synchronous Start/Stop

If you use a MME, WDM-Audio or DirectSound application program with more than one recording or playback device of the MARC X, you will have to determine, whether you want to operate these devices synchronously or independently from each other.

"Independent Start/Stop" allows an independent start/stop of the devices, i.e. assuming to have separate sound cards. Most radio automation programs and also DJ programs require this setting.

"Synchronous Start/Stop" guarantees that all used devices are started and stopped sample-synchronously. This is required for multi-channel projects of most hard-disk recording programs.

"SyncBus synchronous Start/Stop" guarantees that all used devices are started and stopped sample-synchronously, even if they are positioned on different MARIAN sound systems, which are connected by the SyncBus. This is required for multi-channel projects of most hard-disk recording programs, if you use devices of several MARIAN cards.

If you work with an ASIO application program, then these settings are automatically performed by the ASIO drivers of the MARC X.

7.2.6. Clock and Sample Rate Conflicts

Based on the samples described in chapter "7.2.2 Clock Settings" it is likely that it may lead to situations, in which certain request to the MARC X cannot be served. Examples:

- You already use the internal clock for recording or playback and also try to use the S/PDIF or ADAT input. It's possible that a clock switch-over is necessary, which however would "interfere" a running recording or playback. In this case the MARC X drivers will refuse the usage of S/PDIF or ADAT inputs.
- You use a MARC X device with a certain sample rate and you want to you use one more device with a differing sample rate. In this case the drivers will also refuse the last mentioned, because only one sample rate at a time can be active.
- The MARC X is configured as clock slave and consequently operates synchronously with an external clock source. If you try to use a device with a sample rate which differs from the external sample rate, the driver will refuse the usage of this device.

If one of the above-mentioned or a comparable situation occurs, then the driver will inform you with a suitable warning or error message. Often this is obstructive, because some application programs and the Microsoft DirectX test all available recording and playback devices at start-up, which subsequently leads to an accumulation of such conflicts. For quite some time you would be busy to confirm these messages of the driver by clicking "OK". This is the reason why these error message are disabled after installation.

If you have problems when using recording or playback devices and you want to see their causes, the error message by switching off the option "*Deactivate error message caused by clock and sample rate conflicts*"). Warning and error message can be quite useful when searching for problem causes.

7.2.7. Setting up the driver interfaces in Windows™ 95/98/ME



Here you can determine the settings for application programs which use DirectSound or the TASCAM GigaSampler Interface (GSIF) in Windows™ 95/98/ME.

7.2.7.1. Section *"Direct Sound"*

In this section you determine the settings for applications, which use DirectSound as driver interface.

Here are some information in advance: the ideal settings for the DirectSound usage depend on some factors in your PC. Because of this we cannot recommend any ideal settings, but can provide you some information on what's behind the possibilities for the settings. You will have to test yourself, what settings will suit best. Anyway, using the standard settings will ensure a trouble-free operation.

"Adjust Latency Manually"

Moving the time control changes the size of the audio buffer on the card. 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 does not sound correct, 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 Manager and the real time conversion is inapplicable.

"Available Playback Devices limited to:"

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

7.2.7.2. Section "GigaSampler/GigaStudio"

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 the data blocks will have to be, 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 leads more easily to interruptions during playback.

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

7.2.8. Setting up the driver interfaces in Windows™ 2000/XP

7.2.8.1. MME and DirectSound

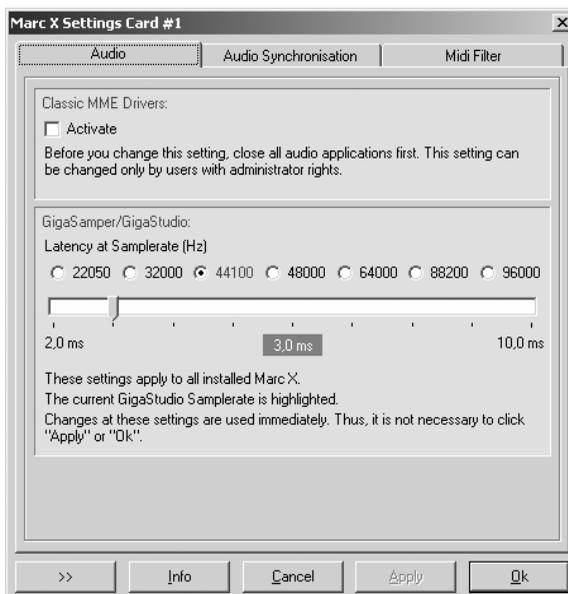
Audio applications, who do not use ASIO, GSIF or direct WDM-Audio (like Cakewalk Sonar), communicate with Microsoft MME or Microsoft DirectSound drivers instead of communicating with the MARIAN drivers directly. The Microsoft drivers in turn communicate with the MARIAN WDM-Audio drivers.

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:
 - under Windows 2000 to 10 devices each (20 channels)
 - under 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)" or "(5+6)". This is caused by an anomaly of the Microsoft MME/DirectSound system. Please ignore these devices and do not use them.
- 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.

7.2.8.2. 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 X 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)

7.2.8.3. Section “GigaSampler/GigaStudio”

In section GigaSampler/GigaStudio you can setup the latency between a Tascam GigaStudio MIDI event and the resulting sound playback.

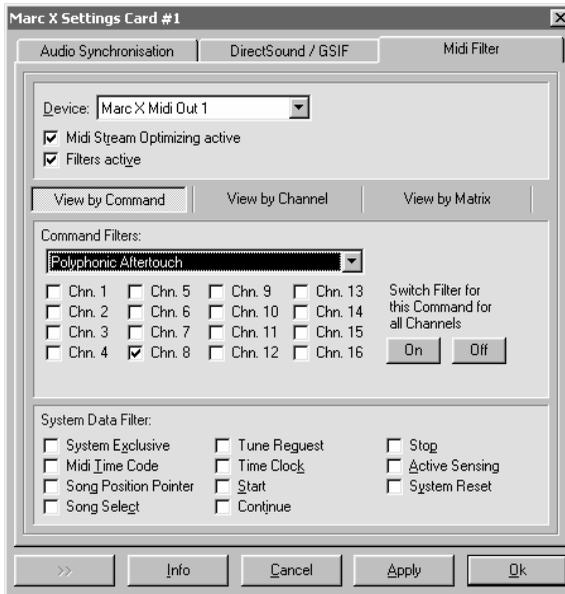
Move the latency slider right, to increase the latency value. Move the latency slider left, to decrease the latency value.

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.

7.2.9. MIDI Settings

In this tab you can edit and especially filter all MIDI commands processed by the MARC X.



"Device"

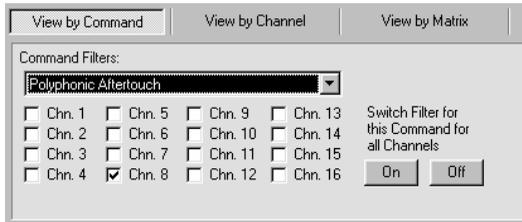
Here you can select the MIDI input or MIDI output for the command, which you wish to change.

"MIDI stream optimization active"

When enabling this option, you filter all unnecessary information from the transmitted MIDI commands. This way you achieve shorter processing times.

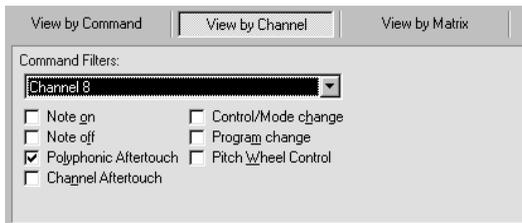
7.2.9.1. Section "Command Filters"

In the Command Filter section channel related commands are processed. There are three view modes available: "**View by command**", "**View by channel**" and "**View by matrix**".

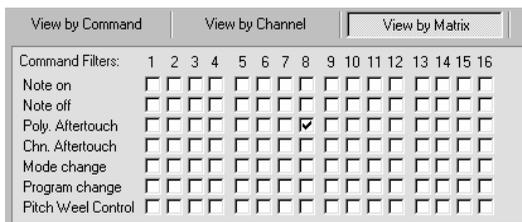


The related level for the "**View by command**" are the individual MIDI commands. This gives you the possibility to filter or transmit a command on the individual MIDI channels. Using the "On" and "Off" buttons you can filter or transmit the command.

Please note, that enabling a checkbox means setting the appropriate filter, i.e. the non-transmission of the MIDI command.



The related level of the "**View by channel**" are the individual MIDI channels and provides the filtering or transmission of the individual commands for the selected channel.



The "**View by matrix**" shows all settings in an overview. The columns show the 16 MIDI channels and the rows show the MIDI commands.

7.2.9.2. Section "System Data Filters"

System Data Filter:

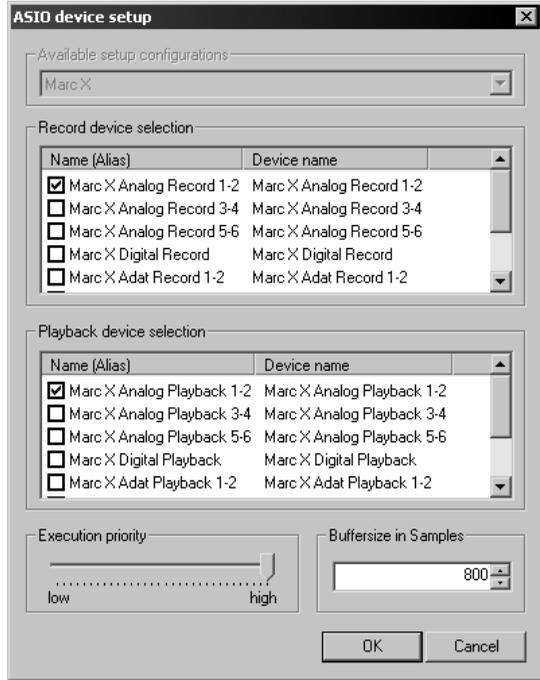
<input type="checkbox"/> System Exclusive	<input type="checkbox"/> Tune Request	<input type="checkbox"/> Stop
<input type="checkbox"/> Midi Time Code	<input type="checkbox"/> Time Clock	<input type="checkbox"/> Active Sensing
<input type="checkbox"/> Song Position Pointer	<input type="checkbox"/> Start	<input type="checkbox"/> System Reset
<input type="checkbox"/> Song Select	<input type="checkbox"/> Continue	

In this section you can filter the channel independent MIDI commands like system commands. All performed settings also apply for the selected MIDI device.

8. ASIO Settings

The picture on the right shows the ASIO controls panel of the MARC X 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 your project! Each activated device reserves system resources and CPU load – even if you do not use the device in your audio project.

Please note that additional settings are necessary to use all selected devices within your ASIO application. For details please refer to the manual of your application program.

If you activate an ADAT or S/PDIF recording device, please ensure that the according input signal is based on the clock source which you choose within your ASIO application. Only in this case a proper recording is possible. Please see chapter "7.2.2 Clock Settings" for more details.

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 X 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 X Analog 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".

10. Technical Data

- Stereo Analog I/O (1/4" TRS jack)
- Stereo Digital I/O S/PDIF (RCA and TOSLINK)
- 8-channel Digital I/O ADAT (switch able to TOSLINK)
- Audio Connector for internal Digital Audio CD/DVD drive
- Audio Connector for internal Analog Audio CD/DVD drive
- Sample formats: 8, 16, 20, 24, 32 Bit Mono/Stereo
- Sample rates Analog: 8 - 96 kHz +/- 15% Pitch
- Sample rates Digital: 32 - 96 kHz +/- 15% Pitch
- Sample rates ADAT: 32 - 48 kHz +/- 15% Pitch
- Frequency response @ 44,1 kHz: 20 Hz to 20 kHz
- Frequency response @ 96 kHz: 20 Hz to 40 kHz
- Maximum input and output level: +8 dBu
- Signal-to-noise ratio S (N+D) AD: 104 dB(A)
- Signal-to-noise ratio S (N+D) DA: 114 dB(A)
- THD: 0,005%

Included Wordclock/MIDI extender board „MWX“

- 2 MIDI I/O via adapter cable
- Wordclock I/O, Superclock I/O (BNC)

Optional Analog I/O extender board „ANX“

- 2 Stereo Analog I/O (1/4" TRS jack)

Optional AES/EBU extender board „DGX“

- 2 AES/EBU I/O (XLR via breakout cable)
- Output voltage $3V_{ss}$ on 110 Ohm
- Output sample rate: 8 - 108 kHz
- Input sensibility approx. $200mV_{ss}$
- Input impedance: 110 Ohm
- Input sample rate: 32 - 96 kHz
- Input sample rate converters:
 - Input SNR: 128dB
 - Input THD+N : 0,00032%
 - Max. sample rate ratio In/Out: 1:3 resp. 3:1

11. Service and Support

If you have any questions or problems during the installation or operation of your MARC X 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. Have a look in our Audio Guide, which has been published on the internet, whether you can find a solution to your problem or an answer to your question. Here you also find the *Audio Guide*:
www.marian.de
3. 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.

You can contact our Sales Department on the internet at
www.marian.de or sales@marian.de.