

IF YOU JUST CAN'T WAIT...

to hear the M5000 AUDIO MAINFRAME, we understand.

If you're not particularly familiar with this type of product please follow the directions below EXACTLY as given and get ready to hear some great sounding effects.

- #1 Make sure that the M5000 is unplugged, then patch the rear panel analog LEFT and RIGHT outputs to the inputs of a high quality amplification system. Turn the volume control on your amplifier fully counter clockwise.
- #2 Connect a high quality signal source, e.g. a CD-player, to the LEFT and RIGHT analog inputs.
- #3 Check that the rear panel voltage selector is set to the correct voltage in your part of the world (only older M5000 models) and plug the line cord into an AC outlet.
- #4 Turn the power on. The LCD display will read out the current software version. After a few seconds the relays will activate and the M5000 is ready.
- #5 The red LEDs in the device selector placed over the disk drive slot, will indicate which ADA-1, DSP-1 or DSP-2 is active. In order to get a clean distortion-free sound, the input levels must be set no higher than the yellow -3 dB LED on each channel just flickers with the maximum signal. The red light indicates DSP clipping (-1 dB). To set the input level refer to the "UTILITY HANDLING" module in the "GENERAL INSTRUCTIONS" section.
- #6 Now, turn up the volume and you should hear the excellent sound of the M5000 mixed with your source-signal. By turning the PROGRAM dial you can choose another factory program. To activate the new factory program, press the DO-button. Refer to "PROGRAM HANDLING" module in the "GENERAL INSTRUCTION" section.
- #7 After pressing the EDIT-button you can modify the current factory program just by turning the "soft dials" A to D. By pressing the PAGE-buttons forward or backward you gain access to all the parameter. You will find a detailed description of the parameters in the "BASIC ALGORITHM" section. Feel free to experiment with them. It is an easy and good way to learn.

By now you should be pretty impressed, but there's much more to come... **so keep reading, trying and experimenting, and you'll soon be an expert.** Don't hesitate to try out the "GUIDED TOURS" section - another easy way to learn the M5000.

WELCOME

M5IFYOU

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INTRODUCTION

This section contains an introduction to the mainframe concept as this idea may be new to some people in the audio business. It is also an introduction to this manual, which is built up in text modules in order to be updated easily. If you later receive a manual module, insert it in the right place and check the appropriate box. This section contains the following text modules:



THE MAINFRAME CONCEPT



HOW TO USE THIS MANUAL

MAINFR. INTRO
M5INTRO
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THE M5000 MAINFRAME CONCEPT

WHY A MAINFRAME ?

The purpose of the mainframe concept is the flexibility to keep up with the ever evolving state of technology. New inventions developed because of the advancing needs of professional engineers can be implemented in the mainframe without the need to scrap a valued piece of equipment. Furthermore, only one user interface is needed to control several modules, i.e. the front panel of the ATAC is controlling one module at the time, although all modules installed are active. (Shown on fig. 1 is M5000).

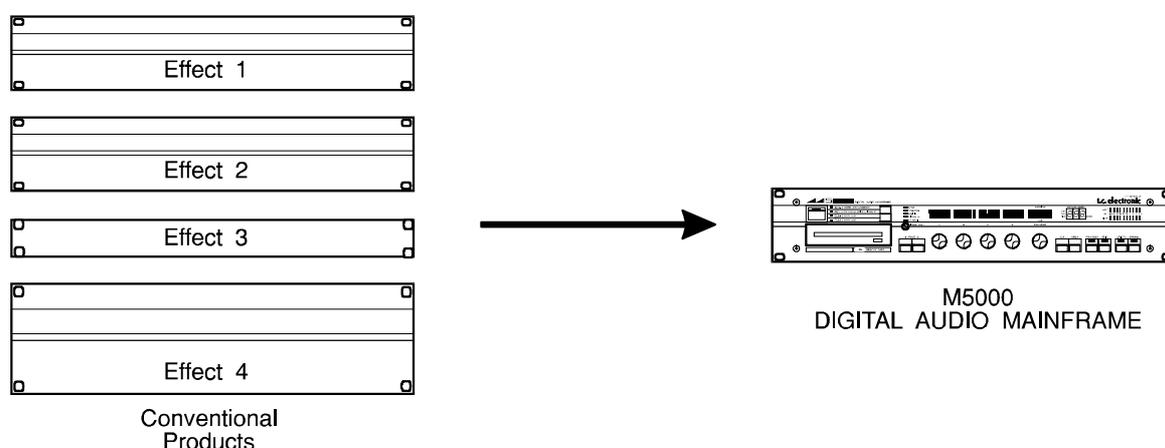


Fig. 1

HARDWARE

The modules are installed in 4 slots placed on the rear panel. Each module is held in with only 2 screws which makes replacement of hardware for updates and upgrades very easy. Once placed into the slot the module is connected to a high speed **24 bit** audio bus. 24 bit makes the M5000 well prepared for the future as 24 bit conversion is not likely to be exceeded for many years to come. The AD/DA converter features 18 bit resolution 64 times oversampling in, 20 bit out. The 24 bit audio bus features up to 64 audio channels which makes the M5000 "patchable" for almost any purpose (fig. 2).

SOFTWARE

As the M5000 is totally software controlled it is essential that also the software is easily exchanged. New algorithms and programs will be available to you in different categories. Some of the programs will be created by other M5000 users and will be available as public domain software, i.e. users can share programs/sounds for free. Other programs and algorithms will be created by well known engineers and musicians. There is more information

on the User Registration Form which we highly recommend that you return to the nearest TC office or to the head office in Denmark.

New software will be available in 2 ways: Floppy Disk or Memory Card. Moreover, software can be loaded into the M5000 via MIDI from another M5000 or from a computer with a MIDI interface (IBM™ compatible, Macintosh™ or Atari™). An electronic bulletin board has been established at TC's head office, so if you have a modem connected to your computer you can get the latest software and information on the M5000. Here you can download the necessary software for dumping software from your computer to the M5000. More on this in APPENDIX F.

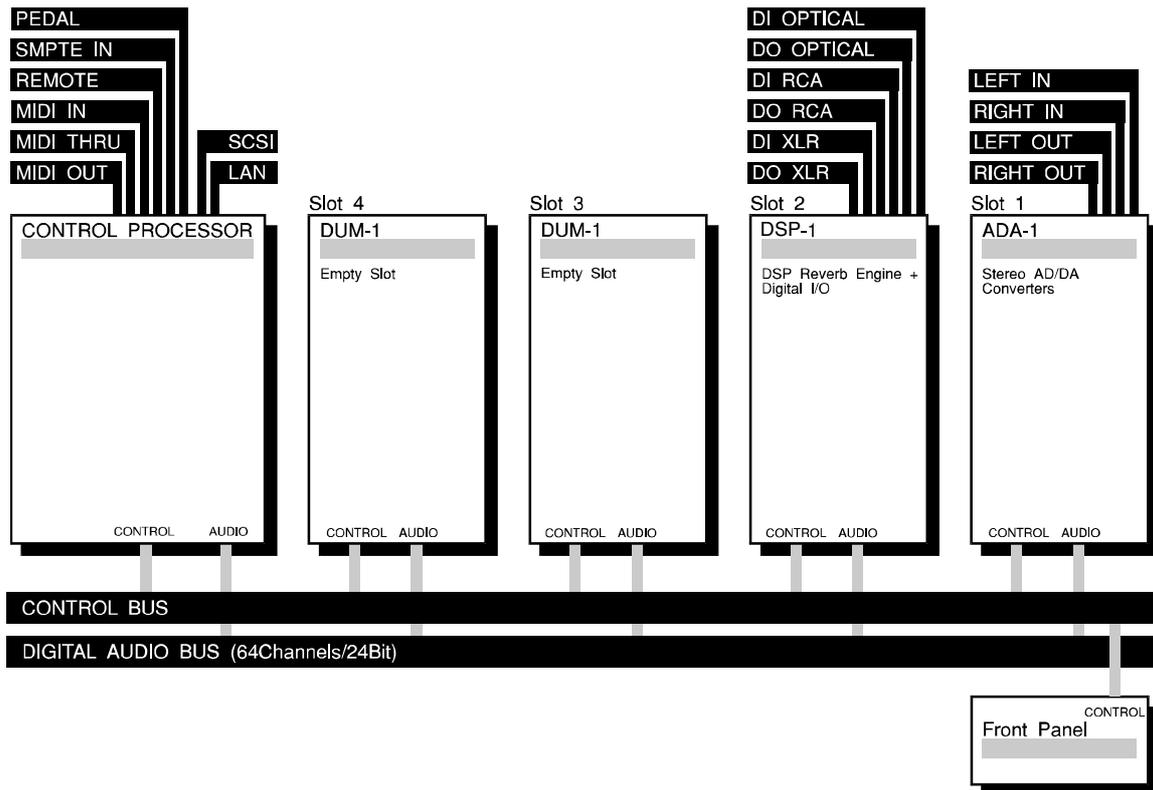


Fig. 2

FLOPPY DISK

The floppy disk is IBM™ compatible and can be copied on any IBM/clone PC. You can borrow a colleague's disk if he should have the latest software version and transfer this software into the permanent memory of the M5000. If your colleague has been charged for this software the M5000 will prompt you for a unique access code in order to install the software in your machine. As soon as you have the access code your M5000 will accept the new software installation. So, where do you get the access code? You call your dealer or TC sales office and on basis of information provided by you (serial number, software type, payment method, etc.) this access code is generated. More information about this system will

be provided to registered M5000 owners as the first chargeable software updates become available.

MEMORY CARD

Software updates may be installed just as easy using the memory card. The memory card is a credit card sized card which exists in many different types and capacities. The M5000 supports the JEIDA, PCMCIA type card, a world-wide portable computer standard.

A freelance engineer often works on different gear. In this situation it is easy to bring his own programs stored on the memory card and install them in the different M5000s without overwriting the existing programs. This is more thoroughly explained in the "PROGRAM HANDLING"-module in GENERAL INSTRUCTION SECTION.

SOFT/HARDWARE FUTURE OPTIONS

When new software packages are released, they will allow the M5000 owners to listen and try these for a limited active time - **free of charge**, thus enabling the user to hear and try the new sounds on your own machine, before purchase !

New hardware modules are also planned in the future.

GENERAL INSTRUCTIONS

This section contains description of the general functions and procedures for the mainframe, no matter what hardware or software configuration you have. As the user interface is the same for all algorithms installed the global parameters are described in this section. The section contains the following text modules:

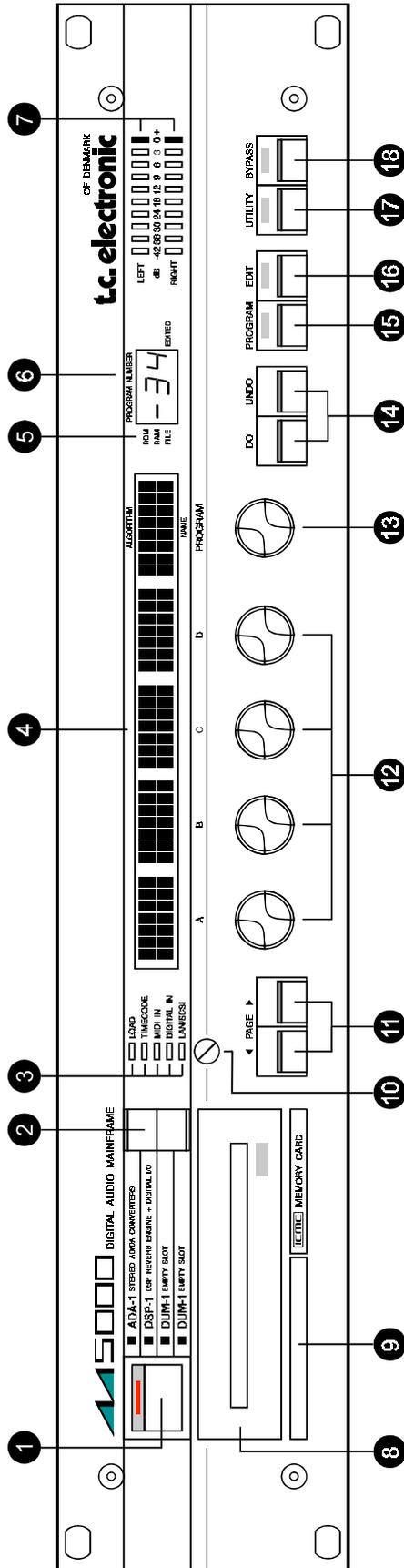
- FRONT/BACK PANEL DESCRIPTION
- PROGRAM HANDLING
- DISK/CARD HANDLING
- UTILITY HANDLING

GENERAL INSTRUCTIONS

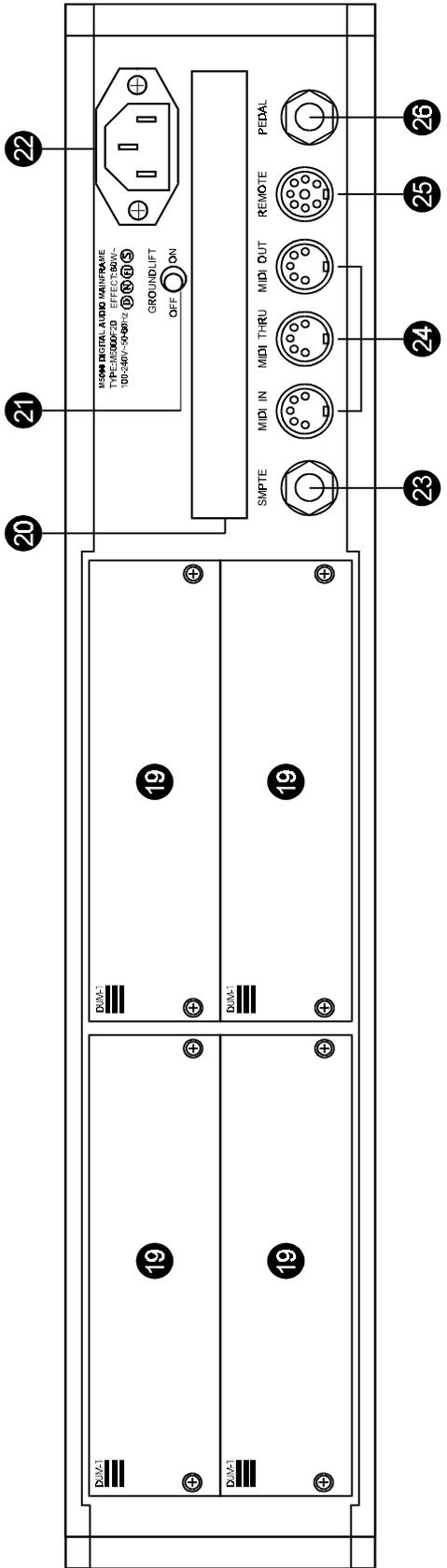
FRONT PANEL

M5000

- 1. POWER SWITCH** The main power On/Off switch.
- 2. DEVICE SELECTOR** Selects between the installed kits to be operated from the mainframe front panel controls. The LED's and labels correspond to the configured kits.
- 3. LOAD LED** Lit while parameters/programs are being updated
TIMECODE LED Lit when receiving timecode
MIDI IN LED Lit when receiving MIDI
DIGITAL IN Lit when receiving at digital inputs and SAMRATE is locked
LAN/SCSI Lit when reading and writing data
- 4. DISPLAY** 80 character alphanumeric display. The top line is divided into 5 sections and tells which 4 parameters and which algorithm is currently being modified. The bottom line is dedicated to the five "soft dials" that are directly below it and displays the 4 parameter values and the program name.
- 5. ROM, RAM, FILE** ROM indicates that the factory program bank is selected for RECALL, STORE or PREVIEW. RAM indicates that the user program bank is selected. FILE indicates that programs relate to an external file on memory card or floppy disk.
- 6. PROGRAM NUMBER** Shows either the origin of the current setting or, if blinking, the current previewed program. If the current setting has been edited the small EDITED LED will be lit.
- 7. LEVEL METER** Dual 10 segment LED bargraph. Displays the input or output level on the DSP-module. Red light indicates **DSP** overload.
- 8. DISK DRIVE** Disk drive for updating software and storage of programs. The M5000 can be updated with new algorithms through the disk drive.



- 9. MEMORY CARD SLOT** Load and stores programs into the M5000. Makes transfer of "personal" programs from one M5000 to another very easy by means of the "credit card"-like memory card.
- 10. VIEWING ANGLE** Makes the alphanumeric display readable at almost any angle.
- 11. PAGE BUTTONS** As there are more parameters to edit than possible to show at the same time the page buttons scrolls through the parameters on the display.
- 12. SOFT DIALS A, B, C, D** Used for editing the parameter values on the display just above.
- 13. PROGRAM DIAL** Turn this dial to preview the programs. Also used when recalling, storing and renaming programs. The top line on the display shows the current algorithm type. The bottom line shows the name of the program.
- 14. DO, UNDO BUTTONS** When operating the M5000 many functions will not be executed until DO is pressed, e.g. turning the program dial will not execute the program until DO is pressed. The UNDO button enables you to compare edited program with the original.
- 15. PROGRAM BUTTON** Press this button to select program source, store or rename programs.
- 16. EDIT BUTTON** Press this button to edit the parameters in the current program. As soon as a stored program is edited the small 'edited' LED in the program number display will be lit until it is stored again.
- 17. UTILITY BUTTON** The UTILITY mode enables the user to access various utility menus for setting the selected kit.
- 18. BYPASS BUTTON** Press this button to bypass the current kit controlled by the mainframe, corresponding to the selected devices (2).



GEN. INSTR.
M5GENIN
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- 19. MODULE SLOTS** This is where the M5000 module cards are installed. With four module slots the M5000 frame can house for example 4 full stereo reverbs modules with digital I/Os.
- 20. OPTION** Future OPTION such as SCSI, PCMCIA and other future expansions may be configured to this port.
- 21. GROUNDLIFT** In position OFF : Direct connection from internal ground to chassis. In position ON : Internal ground connection to chassis through a capacitor. Also called 'flying chassis'.
- 22. AC CONNECTOR** Connector for AC power cord. 3 prong IEC type. The center post is chassis ground. Input voltage : 100-240 Vac, 50-60 Hz.
- 23. SMPTE INPUT** Enables the M5000 to make program changes and other pre-programmed functions as it is synchronized to timecode. Refer to the "MIDI & SMPTE" section for more information.
- 24. MIDI CONNECTORS** MIDI data can be read and generated from these connectors. MIDI THRU sends a duplicate copy of the data received at MIDI IN.
- 25. REMOTE** Connects to the ATAC remote controller. The port communicates with the remote through bi-directional serial data transmission.
- 26. PEDAL CONNECTOR** Used for a simple external switch. The function of the switch will be programmable.

Concept

The M5000 handles 3 different program sources: ROM, RAM and FILE (fig. 1).

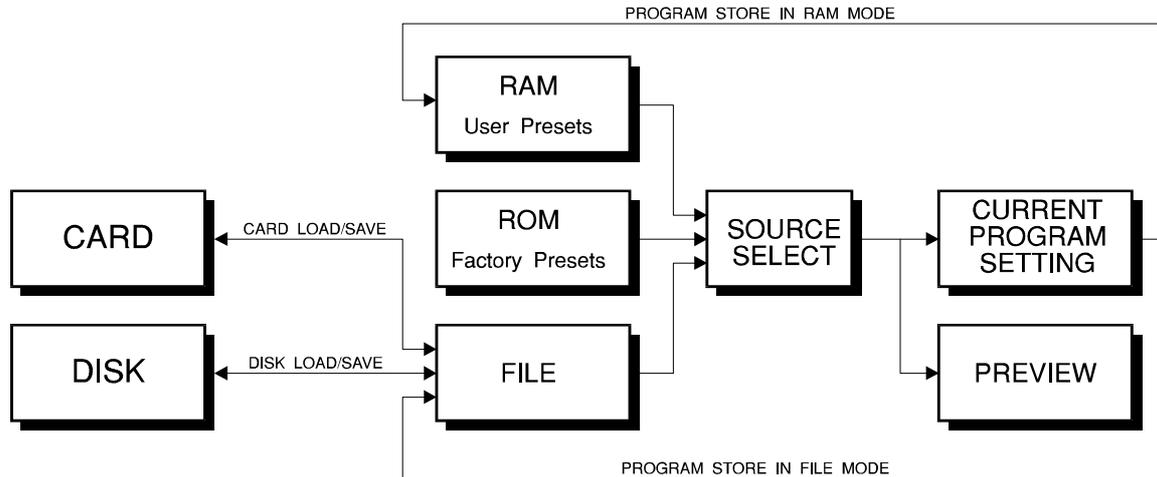


Fig. 1

- ROM** In ROM (Read Only Memory) you will find the factory programs. Along with the basic operating software there are factory programs implemented and they can not be overwritten, i.e. programs can not be stored back in ROM. The factory programs will increase as TC DSP programmers are continuously working on new programs. In the future these programs will be available either **free of charge** or for a moderate fee.
- RAM** In RAM (Random Access Memory) you are able to store an edited program, i.e. here you will find the so-called user programs. This is also where you can build up your own bank of maximum 100 user programs for instant recall. A long life lithium battery keeps these programs in RAM after power down.
- FILE** The FILE buffer is where programs are loaded from and stored to external devices. External devices can be either floppy disk or memory card. As in RAM the programs loaded into the FILE buffer can be recalled, edited and then stored again. One main difference from RAM is that programs in the FILE buffer are **not backed** up by battery and therefore will be lost after a power down. Before power down the FILE buffer **must be saved** to either floppy disk or memory card if you want to keep your presets. You may also copy the FILE buffer in to RAM, overwriting the existing presets.

When a program is recalled - either selected from ROM, RAM or FILE with the source selector (16A, in program mode) - it is **copied** in to a working memory. Here a program can be edited and it is referred to as the "current setting". This means that it is the current program setting, which is displayed on the alpha-numeric display (4). The current setting is - as RAM - also backed up by battery and will remain in working memory after power down. A small LED in front of the program number in the PROGRAM NUMBER display (6) will indicate where the current program is copied from. As soon as the current setting is edited the small EDITED LED in the PROGRAM NUMBER display (6) will light up. When you wish to store the new edited program you have different options to choose from. You can store it in RAM or FILE and overwrite the original recalled program or you can rename it - creating a new program. Storage means that the current setting in the working memory is copied into either RAM or FILE buffer. All programs can instantly be copied to and from RAM and FILE buffer (fig. 2).

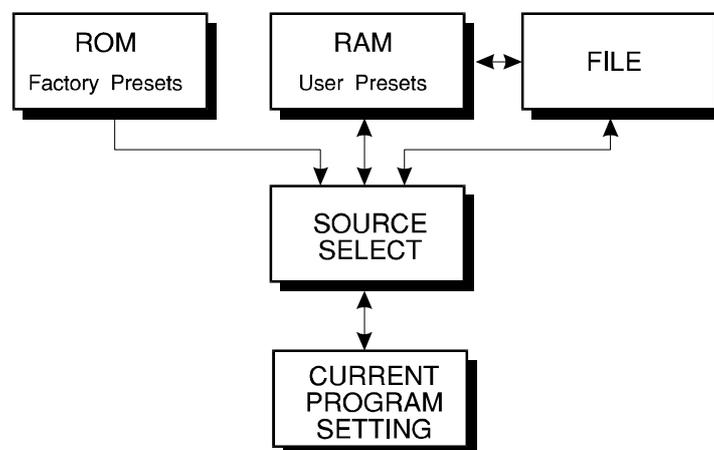


Fig. 2

The purpose of the FILE buffer is that it enables a freelance engineer to bring his own programs on a floppy disk or his "personal" memory card and load them into the FILE buffer without overwriting the M5000's existing RAM programs. He will always know exactly what he is working with as he is familiar with his own programs and doesn't have to look for appropriate programs in RAM. Another use is to stack user modified programs in the FILE buffer for certain recording projects and then save it on a disk dedicated to the specific project. The disk can then be stored with the multitrack tape, sequence disk etc. For a later remix situation the programs can instantly be recalled along with the dedicated sequencer song and sample library.

PROGRAM PARAMETERS:

Pressing the PROGRAM button (15) makes the following parameters appear on the display (4):

PAGE 1 PARAMETERS: RECALL AND STORAGE OF PROGRAMS

<u>CONTROL</u>	<u>TOP LINE</u>	<u>BOTTOM LINE</u>	<u>DEF. VALUE</u>
SOFT DIAL A:	SOURCE SELECTOR	ROM, RAM, FILE	ROM
SOFT DIAL D:	MODE	RECALL, STORE	RECALL
PROGRAM DIAL:	ALGORITHM NAME	PROGRAM NAME	-

PAGE 2 PARAMETERS: CREATE AND RENAME PROGRAM

On the bottom line - over the PROGRAM DIAL (13) - the name of the last recalled program is displayed. If the EDITED LED in PROGRAM NUMBER display (6) is lit this program has been edited and is altered from the original. The edited program can be stored as it is, overwriting the original program under the same name, or stored in another program location in either RAM or FILE. The program can be renamed on page 2. The name of the current program is changed (in either RAM or FILE), not only the name of the current setting. The procedure will not affect the sound.

On the top line the original program name is displayed with a cursor under the first character - ready for editing.

SOFT DIAL A:	CURSOR	Moves the cursor forward or backwards through the name. The name can have a maximum of 8 characters.
SOFT DIAL B:	LETTERS	Selects a letter from A to z and inserts it in the name over the cursor.
SOFT DIAL C:	FIGURES	As letters but numerical from 0 to 9.
SOFT DIAL D:	SYMBOLS	Inserts symbols instead of characters, e.g. blank or space is a symbol found here.
PROGRAM DIAL:	PROGRAM NAME	Shows the original program name. (Press DO to confirm program change/rename)

PAGE 3 PARAMETERS: FILE BUFFER HANDLING

Before any FILE buffer handling is possible one must be created. On page 3, turn soft dial A until "New" occurs. Press the DO button and you have created an empty FILE buffer. You can also simply go directly to Load either Disk or Card. The following parameters are accessible with soft dial A after a File buffer has been created:

New	Creates a new FILE buffer.
Ram To File	Copies all RAM programs to FILE buffer.

Load Disk	Loads programs from floppy disk into the FILE buffer.
Load Card	Loads programs from memory card into the FILE buffer.
Save Disk, Save Card	Saves existing programs in FILE buffer on either floppy disk or memory card. Press DO and a new page will appear. It enables you to create or rename a file. File names are edited as program names. Program dial selects the saving destination, i.e. floppy disk or memory card.
File To Ram	Copies the whole FILE buffer to RAM. Existing RAM programs will be overwritten. Confirm by pressing the DO button.

COMPARE A PROGRAM (UNDO-button)

In order to be able to compare program changes to the original program TC has implemented this A/B-test feature. As mentioned before once the current program setting has been changed from the original program, the small EDITED-LED in the PROGRAM NUMBER display will lit. By pressing the DO-button once, the original program will be loaded again and the EDITED-LED is off. Press the UNDO-button and the previous changed setting is recovered. Switch between original and changed program settings by pressing respectively DO- and the UNDO-button.

The following parameters appears when pressing the UTILITY button (17). The PROGRAM dial (13) selects the menus.

PROGRAM DIAL: **MENU** I/O, G-LEVELS, MIDI, METERS, FILE, PEDAL, ATAC SETUP, CONFIG and SMPTE.

I/O MENU:

PAGE 1:

SOFT DIAL A: **I/O** Selects different input and output configurations. See the following examples:

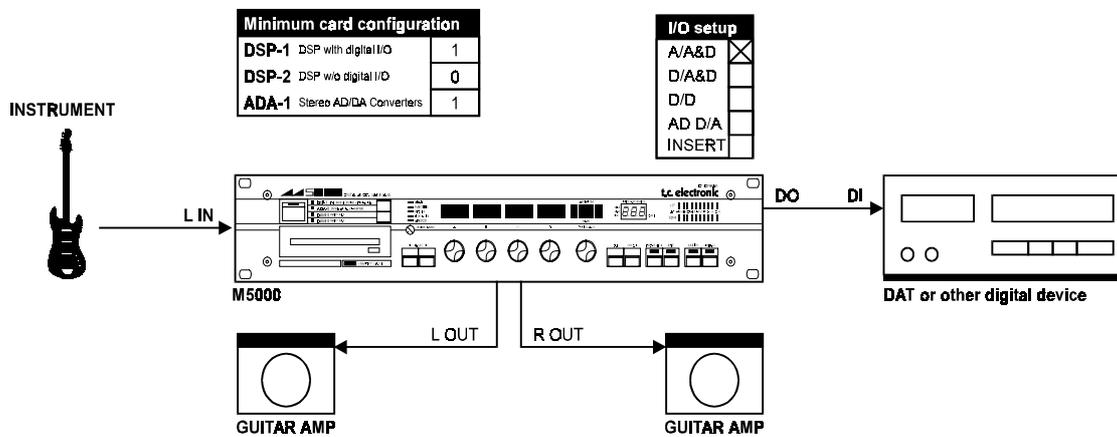


FIG. 1. A/A&D: Analog input and both analog and digital output.

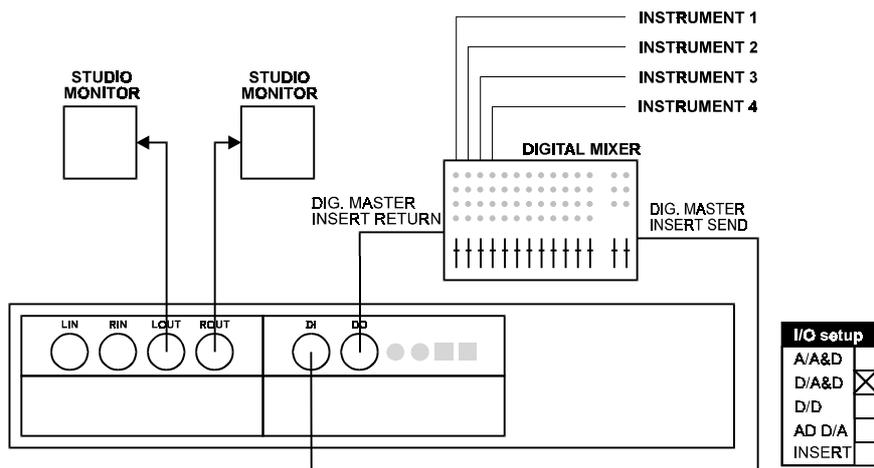


FIG. 2. D/A&D: Digital input and both analog and digital output.

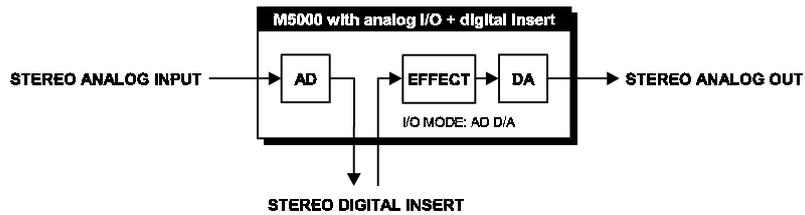


FIG. 3. AD D/A: Analog in and unprocessed digital out simultaneously with digital in and processed signal analog out.

This may be used as a converter mode, where you convert from A to D (**unprocessed** digital out) and also from D to A (analog **processed** signal out) simultaneously and independent. You can use this mode for various purposes. An example is to use the M5000 as an AD converter and connect the house clock to the digital input on the M5000. Another possibility is to use more than one DSP-1 in a serial connection (FIG. 4).

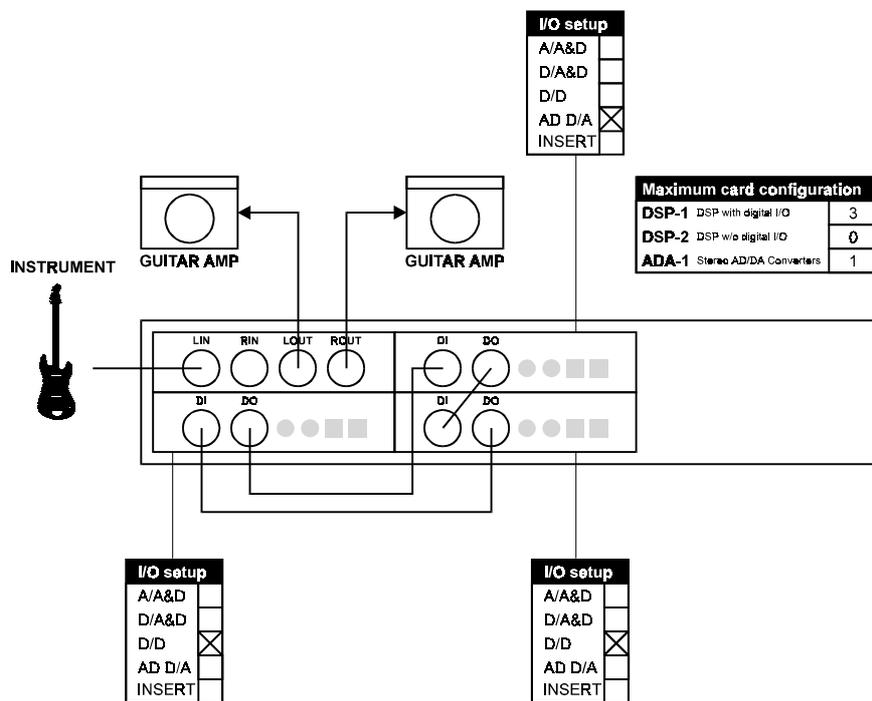


FIG. 4. M5000 as a multieffect unit.

Another example is if you work with a digital storing media and want to record a guitar while you are listening to some other tracks and also on the same time want reverb on these tracks. You can then connect your guitar setup to the analog input on the M5000. From the digital output you connect to the digital input on your DAT/ hard disk (FIG. 5).

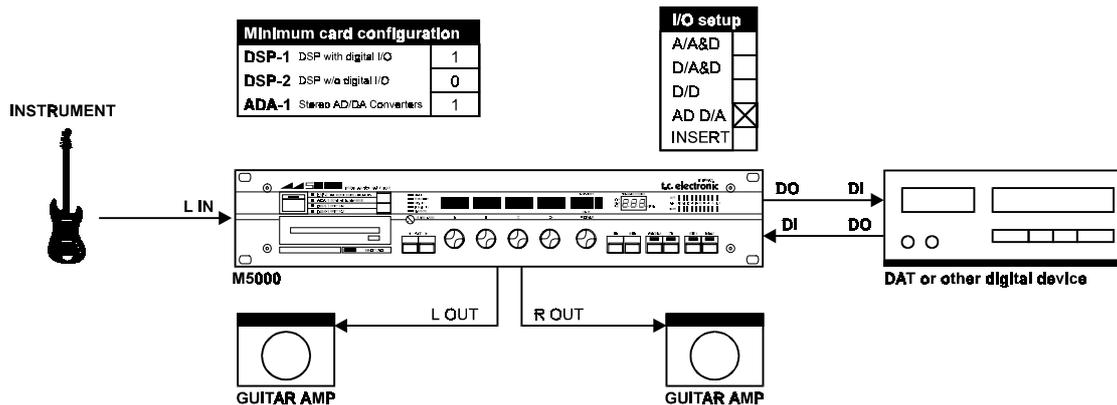


FIG. 5. Digital storing media for instruments.

The guitar signal will be converted for storing on your DAT. At the same time the M5000 can process and convert the tracks selected from your DAT from digital to analog. (You have to connect the digital out from your DAT to the digital in on your M5000).

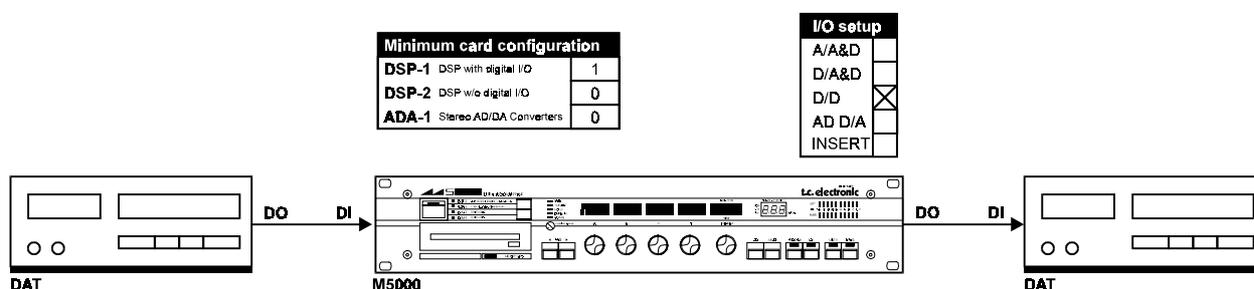


FIG. 6 D/D: Digital input and digital output.

In this mode the M5000 works locally on the DSP-1 card and therefore does not use the bus, which means that you can use a different clock than the one the bus is using.

INSERT

In the TOOLBOX™ algorithm you have the opportunity to insert a second DSP engine internally in the M5000 frame in order to have 2 DSP engines connected into one system (see TOOLBOX™ algorithm description in the BASIC ALGORITHM section). When you are running the TOOLBOX™ on one DSP you can for example run the DYNAMIC1 algorithm on a second DSP engine. The I/O mode on the DSP engine that runs the TOOLBOX™ can be set as normal (see the above options). The second DSP **must be set to INSERT** in order for the DSP to know where to get its signal.

SOFT DIAL B: M-CLOCK

Selection of the master clock. Determines the working speed on the bus. This also means that it is a global parameter that works for all the devices mounted in the frame, except for the devices set to D/D I/O. These devices work locally and can have a work clock different to the one used by the bus.

M5K44.1 &
M5K48.0

This selects the clock that the M5000 works with internally. This clock also functions as the house clock in a digital system (this is of course only if you have DSP-1 with digital I/O.) E.g. if you are working with a system that does not generate a master clock - use either the M5K44.1 or the M5K48.0 setting to make the M5000 generate one of these clocks (44.1 KHz. or 48.0 KHz.) to use as the house clock in your system.

DIN-1, DIN-2,
DIN-3, DIN-4

The M5000 locks on the sample rate from the signal received in either digital input number 1, 2, 3 or 4 according to your selection. DIN-1 corresponds to digital input no. one, DIN-2 to digital input no. 2 and so on. The numbers available depend on the amount of DSP-1s mounted in the frame. If you have one DSP-1 card in your frame you can only select the DIN-1 setting.

SOFT DIAL C: SAMRATE

NONE,
32.0 (LCL)
44.1 (LCL)
48.0 (LCD)

This is a read only indicator. It means that you can not change anything, but only read the sample rate on the display.

This indicator shows the sample rate for the device you have selected to operate. (LCL) after the number means that the device is operating locally and has no contact with the bus (only possible in D/D mode). If the device selected needs to work with the bus (all I/O settings except D/D) the number will correspond to the master clock in the M5000.

SOFT DIAL D: SOURCE STEREO, LEFT, RIGHT, MONO

Input source. Selects between a stereo or mono input signal.

PAGE 2:

SOFT DIAL A: DO-TYPE* S/PDIF, AES/EBU

Determines the digital out type, regardless of the input source used.

SOFT DIAL B: DO-CPY* on, off

Copy protection on/off. The M5000 can remove the copy prohibit bit that is present in the S/PDIF format (DO-CPY=**on**). This means that you can edit your DAT recordings in the S/PDIF format more than once. If DO-copy is **off** the M5000 adds the copy prohibit bit to the digital output.

* Has no influence if DSP-2 is installed.

G-LEVELS MENU:

When an ADA-1 is installed in the M5000, audio levels must be adjusted in order to get a clean distortion-free sound. Once analog inputs and outputs are connected, levels can be set. The output level can be adjusted to provide levels that matches your console or other audio system. Once a digital signal has been recorded it is often hard to adjust the digital level. In this menu it is possible to attenuate the digital signal, digital gain is available only in the optional MD2 algorithm.

PAGE 1:

SOFT DIAL A:	A-IN	-12 dB to +12 dB	Sets analog input level.
SOFT DIAL B:	A-OUT	-18 dB to +12 dB	Sets analog output level.
SOFT DIAL C:	D-IN	off - 0.0 dB	Digital attenuation.
SOFT DIAL D:	MIXMODE	WET+DRY, WET=MAX and WET=MIX	

As all factory programs are programmed with a mixed signal between dry and wet signal for musical instruments (WET+DRY) TC has implemented this feature. When MIXMODE is set to WET=MAX, it automatically sets the mix level on 100 % meaning that no dry signal is coming through to the effect output. Set MIXMODE **WET=MAX** and all direct signals are "killed" regardless of preset mix settings. This application simplifies the use of the M5000 when it is used with mixing consoles. WET=MIX means that the dry signal is killed, but the programmed MIX percentage remains unchanged from the original.

PAGE 2:

DA-DEMP	none, 48KHz, 44.1KHz, 32KHz	Digital to analog de-emphasis. If you need to deemphase a pre-emphased signal, choose the proper samplerate.
ABSPHAS	neg, pos	With this parameter the absolute phase of the output signals can be inverted. It affects both the analog and digital output signals.
R68-LEV	off, on	When set to on the headroom on analog inputs and outputs are changed to 18 dB according to the EBU TECHNICAL RECOMMENDATION R68-1992.
FST.TRG	off, on	When fast trigger chip is mounted this should be set to on .

MIDI MENU:

For more information about the MIDI menu, please refer to the text module "INTRODUCTION TO MIDI OPERATION" in the MIDI & SMPTE section.

METERS MENU:

Turning the PROGRAM dial to the METERS menu enables you to choose whether you want the meter to display digital input or digital output level. Note that the meters (7) **always** displays the digital levels (in/out of the DSP module). Set the input level so that the green -3 dB LED on each channel just flickers at peak levels. The red 0 dB LED will light up when DSP input is overloaded - even if output level meter is selected.

FILE MENU:

When storing programs on either floppy disk or memory card eventually the disk or card will be full. In the FILE MENU it is possible to view and erase files stored on either disk or memory card.

Entering the FILE MENU the display will show <<FILE PRESS DO>>. Use soft dial A and press DO to select the following functions:

CARD DIRECTORY	Scrolls through the files on the PCMCIA memory card. Views not only program files but also application software files. Press DO or UNDO to return.
CARD DELETE FILE	Scrolls through the files on memory card. Views all files placed on the card. Press DO to erase selected file or UNDO to abort this function. Confirm by pressing DO a second time or press UNDO again to cancel.
CARD FORMAT	Memory card formatting. Press DO and select card size with soft dial A. Confirm by pressing DO once again.
DISK DIRECTORY	Scrolls through the files on disk. Views not only program files but also application software files. Press DO or UNDO to return.
DISK DELETE FILE	Scrolls through the files on disk. Views all files placed on the disk. Press DO to erase selected file or UNDO to abort this function. Confirm by pressing DO a second time or press UNDO again to cancel.
DISK FORMAT	Disk formatting. Press DO and select disk size with soft dial A. Confirm by pressing DO a second time or press UNDO again to cancel.

PEDAL MENU:

It is possible to remote bypass the M5000 with an external switch. Connect your pedal/switch to the pedal connector (26) and use the soft dial A to configure the M5000 to your switch.

SOFT DIAL A: **NONE, BYPASS WHEN TOGGLE, BYPASS WHEN PUSHED, BYPASS WHEN RELEASED.**

CONFIGURATION MENU:

Use DIAL A to select the different options:

SHOW CONFIG (Press DO):

This menu shows (read only) the module configuration for the selected device, the size of the index ram (**IDX=**), and the size of the **DRAM** (extra memory for sampling).

IDX=	The index ram can be either 32K (standard) or 64 K. If you mount a HIMEM kit (64K) it will give your DSP card more memory, thus providing longer delay / pre delay times in some of the algorithms. Exactly where and how much is described under the respective algorithms.
DRAM	Shows how much dynamic RAM (for sampling) is mounted.

M5000 OPTION (Press DO to view):

Dialing forward you will find all the options available using this software version. If the option you selected is not installed, the display will appear as follows:

OPTION	TIME	LEVEL
(Option name)	--	off

Beneath 'OPTION' you see the option names. These options can be installed in 2 ways; PERMANENTLY or TEMPORARY, i.e. for a limited number of hours. All options purchased are of course installed permanently, but we invite you to **try an option free of charge** for a limited time period, usually 100 hours. The installation procedure is described in CONFIGURATION section, OPTION INSTALLATION. Please refer to this text module for information about **how to order a temporary demo option**. The LEVEL parameter describes how many DSP cards in the frame that can run the option simultaneously.

On temporary installations 'TIME' describes the number of hours the option is still available. On permanently (purchased) installed options the time limit will of course be '**forever**'.

SMPTE MENU:

For more information about the SMPTE menu, please refer to the text module "SMPTE OPERATION" in the MIDI & SMPTE section.

THE BASIC ALGORITHMS

Top DSP programmers around the world are working on enhancements to the basic algorithms and are developing new algorithms to achieve a wide range of extraordinary applications.

The M5000 will be up to date for many years as new software packages and hardware modules are developed. The possibilities are virtually unlimited as the M5000 can be configured in many variations for optimal performance for Recording, Broadcasting and Sound Reinforcement.

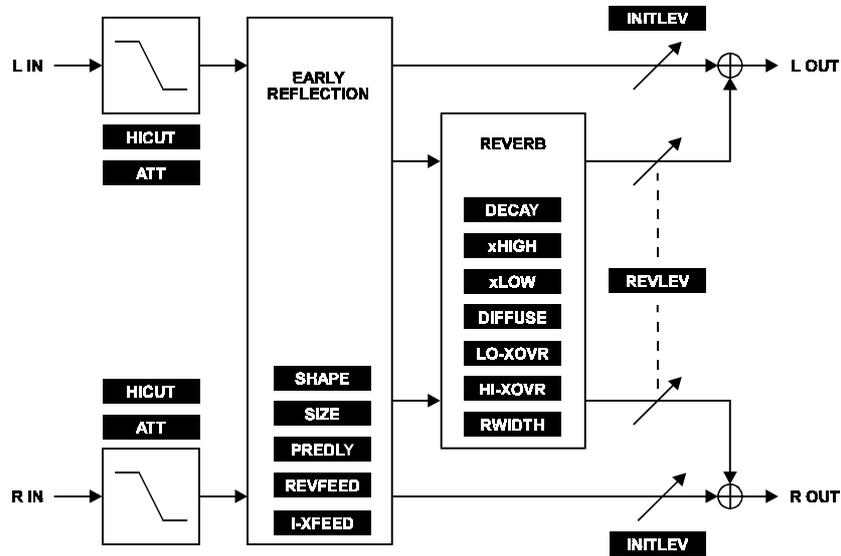
This section will explain the algorithms that come with the software version 2.0 They are as follows:

ROM PRESETS:	
REVERB-1	DELAY-2
REVERB-2	SAMPLE-1
REVERB-3	AMBIENCE
NONLIN-1	TAPFAC
CHORUS-1	PARAM.EQ
REVPITCH	REVCORE-1
PITCH-1	DYNAMIC1
PITCH-2	TOOLBOX
DELAY-1	

On the next page you will find a complete signal flow diagram for the M5000. It shows where the adjustable parameters are placed in relation to the actual algorithm signal flow. The "APPLICATION" box in the middle of the diagram is blank as the text module for each algorithm contains a separate diagram unique to that specific algorithm.

If you receive an update on one of these algorithms in the future you will also receive a new revision of the text module in this manual related to the specific algorithm. Remove the old text module and insert the new one in its place. The latest revision number is marked at the lower right corner of each page with the section name and the module name.

Here is a brief description of the parameters dedicated to the REVERB-1 algorithm. The diagram below is an addition to the signal flow diagram found in the "BASIC ALGORITHMS" module, page 2.



EDIT PARAMETERS:

MIX	0 - 100 %	Sets the mix between dry and wet signal. 100 % = effect signal only. Mix can be set to 100 % globally in the G-LEVELS-menu under UTILITY. Set MIXMODE WET=MAX and all dry signals are "killed" regardless of preset mix settings.
INLEV	off - 0.0 dB.	Sets the level of the input to the reverb in 0.5 dB steps. The function of the control is to maximize dynamic range. Please note that this control is positioned after the input PPM meter, and will not affect the input PPM reading, also if using an analog input, make the analog input adjustment in the G-LEVELS-menu under UTIL before setting this control. However, if the red overload LED flashes, turn down INLEV a couple of dBs. The control does not affect the bypassed signal level. Normally, you do

		not have to change the factory default setting.
OUTLEV	OFF - 0.0 dB.	Sets the output level of the reverb in 0.5 dB steps. The function of this control is to maximize dynamic range by allowing the reverb algorithm to output maximum signal to the DA converters. It affects the output PPM reading. There is a separate output level control for adjusting the analog output level in UTIL. Set the analog input (if you use analog input) and the INLEV adjustments before setting OUTLEV. The control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.
DECAY	0.3 - 60.0 Sec.	Reverberation decay time.
x LOW	0.01 - 2.5 times	Relative decay time multiplier for low frequencies.
x HIGH	0.01 - 2.0 times	Multiplier for the high frequencies. If x HIGH time is set to 0.5, the HI decay time is half that of the nominal DECAY setting.
DIFFUSE	0 - 25	Simulation of reflections in the room "hitting" more or less uneven surfaces. The DIFFUSE parameter affects the density of the reverb tail. To set the DIFFUSE properly, turn off the INITLEV paramter and adjust while listening on percussive type of signals/instruments.
SHAPE	HALL, FAN, PRISM, H.SHOE	Room/Hall simulation/approximation. With this control the initial pattern of the reverb is chosen. In REVERB-1 4 distinctively different room shapes are available. The HALL reflection pattern is based on the acoustic properties of the Boston Symphony Hall, USA . The FAN pattern on a fan-shaped hall akin to the La Scala Concert Hall in Milan, Italy . The PRISM pattern is from acoustic designers 'golden ratio' shoe

box shaped Hall. Finally the Horseshoe shape pattern is based on the **Musikvereinssaal, Austria**. Table 1 shows the actual sizes for the rooms simulated.

M5000 REVERB-1 & 2 algorithms										
		For the HALL pattern:								
SIZE		HALL	FAN	PRISM	H.SHOE	CLUB *	SMALL *	LENGTH	initial delay	revfeed
scale	factor	m3	m3	m3	m3	m3	m3	m	mS	mS
4.000	64	1280000	640000	1024000	896000	320000	128000	153.8	223.60	74.53
3.160	32	640000	320000	512000	448000	160000	64000	122.1	177.48	59.16
2.500	16	320000	160000	256000	224000	80000	32000	96.9	140.86	46.95
2.000	8	160000	80000	128000	112000	40000	16000	76.9	111.80	37.27
1.600	4	80000	40000	64000	56000	20000	8000	61.1	88.74	29.58
1.250	2	40000	20000	32000	28000	10000	4000	48.5	70.43	23.48
1.000	1	20000	10000	16000	14000	5000	2000	38.5	55.90	18.63
0.800	0.5	10000	5000	8000	7000	2500	1000	30.5	44.37	14.79
0.630	0.25	5000	2500	4000	3500	1250	500	24.2	35.22	11.74
0.500	0.125	2500	1250	2000	1750	625	250	19.2	27.95	9.32
0.400	0.0625	1250	625	1000	875	313	125	15.3	22.18	7.39
0.316	0.03125	625	312	500	437	156	62	12.1	17.61	5.87
0.250	0.01563	313	156	250	219	78	31	9.6	13.98	4.66
0.200	0.00781	156	78	125	109	39	16	7.6	11.09	3.70
0.160	0.00391	78	39	62	55	20	8	6.1	8.80	2.93
0.125	0.00195	39	20	31	27	10	4	4.8	6.99	2.33
0.100	0.00098	20	10	16	14	5	2	3.8	5.55	1.85
0.080	0.00049	9.8	4.9	7.8	6.8	2	1	3.0	4.40	1.47
0.063	0.00024	4.9	2.4	3.9	3.4	1	0.5	2.4	3.49	1.16
0.050	0.00012	2.4	1.2	2.0	1.7	1	0.2	1.9	2.77	0.92
0.040	0.00006	1.2	0.6	1.0	0.9	0.3	0.1	1.5	2.20	0.73
*) only in Reverb-2 algorithm										

table 1.

x SIZE	0.040 - 4.000	Scales the dimensions of the simulated space depending on the SHAPE chosen. The specific room that is being simulated is scaled 1:1 at SIZE =1.00. This can then be scaled up or down (see table 1). Provided that the predelay setting is relatively short, the corresponding volume of the simulated space is changed radically with this control. For example; with the HALL initial pattern, the approximate room volume goes from 1.2 cubic meters to 1,280,000 cubic meters (table 1).
PREDLY	0.0 - 200.0 mS or 0.0 - 520.0 mS ¹	Sets the time that passes before the first reflection appear. Maximum predelay depends on SHAPE (see table 2).

¹Only if idx RAM mounted is 64K. If idx=32K max predelay will be 200.0 mS. Check your index ram in the CONFIG menu under UTILITY.

Increasing the predelay will change the apparent position and, to some degree, the size of the room.

Max predelay before loosing taps (std. memory)						
@48KHz samplerate						
	Size	mS	Size	mS	Size	mS
Hall	0.50	112.4	1	40.7	1.28	0.0
Fan	0.50	146.8	1	109.5	2.47	0.0
Prism	0.50	135.4	1	86.7	1.89	0.0
H.Shoe	0.50	116.2	1	48.3	1.36	0.0
Club	0.50	133.4	1	82.7	1.81	0.0
Small	0.50	141.2	1	98.2	2.14	0.0
@44.1KHz samplerate						
	Size	mS	Size	mS	Size	mS
Hall	0.50	118.2	1	52.3	1.40	0.0
Fan	0.50	149.8	1	115.5	2.69	0.0
Prism	0.50	139.4	1	94.6	2.06	0.0
H.Shoe	0.50	121.7	1	59.3	1.48	0.0
Club	0.50	137.5	1	90.9	1.98	0.0
Small	0.50	144.7	1	105.2	2.33	0.0
Max predelay before loosing taps (w.option 'himem' memory, order# 51RAM-1)						
@48KHz samplerate						
	Size	mS	Size	mS	Size	mS
Hall	0.50	459.5	1	393.6	4.00	0.0
Fan	0.50	491.2	1	456.9	4.00	251.2
Prism	0.50	480.7	1	436.0	4.00	167.6
H.Shoe	0.50	463.0	1	400.6	4.00	26.2
Club	0.50	478.8	1	432.2	4.00	152.7
Small	0.50	486.0	1	446.6	4.00	209.9
@44.1KHz samplerate						
	Size	mS	Size	mS	Size	mS
Hall	0.50	500.2	1	428.5	4.00	0.0
Fan	0.50	534.6	1	497.3	4.00	273.4
Prism	0.50	523.2	1	474.5	4.00	182.4
H.Shoe	0.50	504.0	1	436.1	4.00	28.6
Club	0.50	521.2	1	470.5	4.00	166.2
Small	0.50	529.0	1	486.0	4.00	228.5

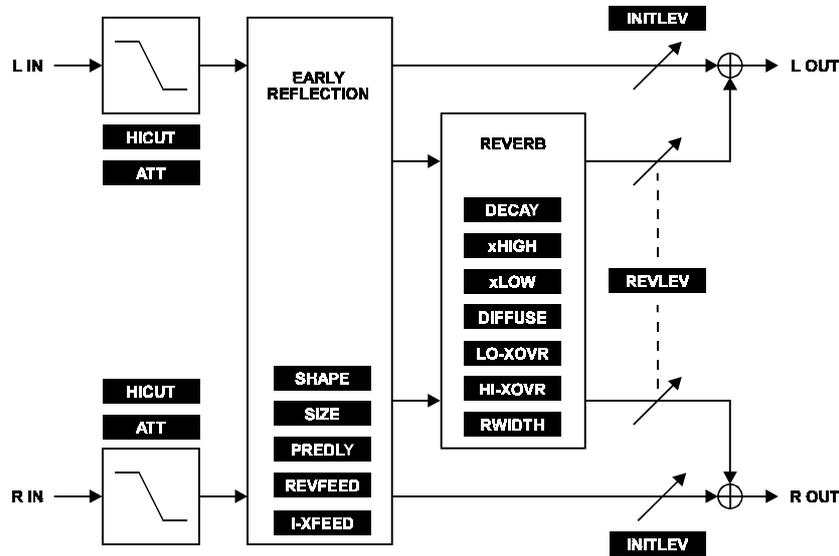
table 2.

REVFEED	0.0 - 100.0 mS 0.0 - 300.0 mS ²	Sets the time before the reverberating part of the signal starts to build up, relative to the early reflections PREDLY.
HICUT	500 Hz - flat	High cut filter, shelving type. Provides an overall reverb high frequency rolloff (6 dB per octave) that is well suited to make a warmer sound. Sets the cut-off

²Only if idx RAM mounted is 64K. Check your index ram in the CONFIG menu under UTILITY.

		frequency of the overall High cut filter in 1/3-octave steps.
ATT	-40 - 0.0 dB	The attenuation control sets the high frequency rolloff determined by HICUT.
LO-XOVR	20 Hz - flat	Sets the crossover frequency for the x LOW decay time multiplier in 1/3-octave steps.
HI-XOVR	20 Hz - flat	Sets the crossover frequency for the x HIGH decay time multiplier in 1/3-octave steps.
INITLEV	off - 0.0 dB.	Sets the level of the initial pattern. The purpose of this control is to balance the initial (early) reflection levels against the reverberating part of the reverb algorithm.
REVLEV	off - 0.0 dB.	Sets the level of the reflection envelope relative to the early reflections in 0.5 dB steps. If REVLEV is set to off you will hear only the initial reflections.
RWIDTH	0 - 100 %	Sets the apparent stereo width of the reverberating part of the algorithm. At '0', the reverb tail will appear to be coming mainly from the center (mono compatible), whereas with RWIDTH set to '100' the L/R reverberators are independent.
I-XFEED	on/off	With this parameter switched off, the cross feeds in the early reflections will be killed. The I-XFEED switched off, simultaneously with the parameter RWIDTH set to 100%, will create a true stereo reverb. The effect from the left and the right channel will be generated totally independent. This is ideal for working with Dolby surround or for broadcasting in general where mono compatibility is important. The feature is also especially applicable for the film industry and post production suites.

Here is a brief description of the parameters dedicated to the REVERB-1 algorithm. The diagram below is an addition to the signal flow diagram found in the "BASIC ALGORITHMS" module, page 2.



EDIT PARAMETERS:

- | | | |
|-------|---------------|--|
| MIX | 0 - 100 % | Sets the mix between dry and wet signal. 100 % = effect signal only. Mix can be set to 100 % globally in the G-LEVELS-menu under UTILITY. Set MIXMODE WET=MAX and all dry signals are "killed" regardless of preset mix settings. |
| INLEV | off - 0.0 dB. | Sets the level of the input to the reverb in 0.5 dB steps. The function of the control is to maximize dynamic range. Please note that this control is positioned after the input PPM meter, and will not affect the input PPM reading, also if using an analog input, make the analog input adjustment in the G-LEVELS-menu under UTIL before setting this control. However, if the red overload LED flashes, turn down INLEV a couple of dBs. The control does not affect the bypassed signal level. Normally, you do |

		not have to change the factory default setting.
OUTLEV	OFF - 0.0 dB.	Sets the output level of the reverb in 0.5 dB steps. The function of this control is to maximize dynamic range by allowing the reverb algorithm to output maximum signal to the DA converters. It affects the output PPM reading. There is a separate output level control for adjusting the analog output level in UTIL. Set the analog input (if you use analog input) and the INLEV adjustments before setting OUTLEV. The control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.
DECAY	0.3 - 60.0 Sec.	Reverberation decay time.
x LOW	0.01 - 2.5 times	Relative decay time multiplier for low frequencies.
x HIGH	0.01 - 2.0 times	Multiplier for the high frequencies. If x HIGH time is set to 0.5, the HI decay time is half that of the nominal DECAY setting.
DIFFUSE	0 - 25	Simulation of reflections in the room "hitting" more or less uneven surfaces. The DIFFUSE parameter affects the density of the reverb tail. To set the DIFFUSE properly, turn off the INITLEV parameter and adjust while listening on percussive type of signals/instruments.
SHAPE	HALL, FAN, PRISM, H.SHOE	Room/Hall simulation/approximation. With this control the initial pattern of the reverb is chosen. In REVERB-1 4 distinctively different room shapes are available. The HALL reflection pattern is based on the acoustic properties of the Boston Symphony Hall, USA . The FAN pattern on a fan-shaped hall akin to the La Scala Concert Hall in Milan, Italy . The PRISM pattern is from acoustic designers 'golden ratio' shoe

box shaped Hall. Finally the Horseshoe shape pattern is based on the **Musikvereinssaal, Austria**. Table 1 shows the actual sizes for the rooms simulated.

M5000 REVERB-1 & 2 algorithms										
		For the HALL pattern:								
SIZE		HALL	FAN	PRISM	H.SHOE	CLUB *	SMALL *	LENGTH	suggested	suggested
scale	factor	m3	m3	m3	m3	m3	m3	m	initial delay	revfeed
									mS	mS
4.000	64	1280000	640000	1024000	896000	320000	128000	153.8	223.60	74.53
3.160	32	640000	320000	512000	448000	160000	64000	122.1	177.48	59.16
2.500	16	320000	160000	256000	224000	80000	32000	96.9	140.86	46.95
2.000	8	160000	80000	128000	112000	40000	16000	76.9	111.80	37.27
1.600	4	80000	40000	64000	56000	20000	8000	61.1	88.74	29.58
1.250	2	40000	20000	32000	28000	10000	4000	48.5	70.43	23.48
1.000	1	20000	10000	16000	14000	5000	2000	38.5	55.90	18.63
0.800	0.5	10000	5000	8000	7000	2500	1000	30.5	44.37	14.79
0.630	0.25	5000	2500	4000	3500	1250	500	24.2	35.22	11.74
0.500	0.125	2500	1250	2000	1750	625	250	19.2	27.95	9.32
0.400	0.0625	1250	625	1000	875	313	125	15.3	22.18	7.39
0.316	0.03125	625	312	500	437	156	62	12.1	17.61	5.87
0.250	0.01563	313	156	250	219	78	31	9.6	13.98	4.66
0.200	0.00781	156	78	125	109	39	16	7.6	11.09	3.70
0.160	0.00391	78	39	62	55	20	8	6.1	8.80	2.93
0.125	0.00195	39	20	31	27	10	4	4.8	6.99	2.33
0.100	0.00098	20	10	16	14	5	2	3.8	5.55	1.85
0.080	0.00049	9.8	4.9	7.8	6.8	2	1	3.0	4.40	1.47
0.063	0.00024	4.9	2.4	3.9	3.4	1	0.5	2.4	3.49	1.16
0.050	0.00012	2.4	1.2	2.0	1.7	1	0.2	1.9	2.77	0.92
0.040	0.00006	1.2	0.6	1.0	0.9	0.3	0.1	1.5	2.20	0.73
*) only in Reverb-2 algorithm										

table 1.

x SIZE

0.040 - 4.000

Scales the dimensions of the simulated space depending on the SHAPE chosen. The specific room that is being simulated is scaled 1:1 at SIZE =1.00. This can then be scaled up or down (see table 1). Provided that the predelay setting is relatively short, the corresponding volume of the simulated space is changed radically with this control. For example; with the HALL initial pattern, the approximate room volume goes from 1.2 cubic meters to 1,280,000 cubic meters (table 1).

PREDLY

0.0 - 200.0 mS or
0.0 - 520.0 mS³

Sets the time that passes before the first reflection appear. Maximum predelay depends on SHAPE (see table 2).

³Only if idx RAM mounted is 64K. If idx=32K max predelay will be 200.0 mS. Check your index ram in the CONFIG menu under UTILITY.

Increasing the predelay will change the apparent position and, to some degree, the size of the room.

Max predelay before loosing taps (std. memory)						
@48KHz samplerate						
	Size	mS	Size	mS	Size	mS
Hall	0.50	112.4	1	40.7	1.28	0.0
Fan	0.50	146.8	1	109.5	2.47	0.0
Prism	0.50	135.4	1	86.7	1.89	0.0
H.Shoe	0.50	116.2	1	48.3	1.36	0.0
Club	0.50	133.4	1	82.7	1.81	0.0
Small	0.50	141.2	1	98.2	2.14	0.0
@44.1KHz samplerate						
	Size	mS	Size	mS	Size	mS
Hall	0.50	118.2	1	52.3	1.40	0.0
Fan	0.50	149.8	1	115.5	2.69	0.0
Prism	0.50	139.4	1	94.6	2.06	0.0
H.Shoe	0.50	121.7	1	59.3	1.48	0.0
Club	0.50	137.5	1	90.9	1.98	0.0
Small	0.50	144.7	1	105.2	2.33	0.0
Max predelay before loosing taps (w.option 'himem' memory, order# 51RAM-1)						
@48KHz samplerate						
	Size	mS	Size	mS	Size	mS
Hall	0.50	459.5	1	393.6	4.00	0.0
Fan	0.50	491.2	1	456.9	4.00	251.2
Prism	0.50	480.7	1	436.0	4.00	167.6
H.Shoe	0.50	463.0	1	400.6	4.00	26.2
Club	0.50	478.8	1	432.2	4.00	152.7
Small	0.50	486.0	1	446.6	4.00	209.9
@44.1KHz samplerate						
	Size	mS	Size	mS	Size	mS
Hall	0.50	500.2	1	428.5	4.00	0.0
Fan	0.50	534.6	1	497.3	4.00	273.4
Prism	0.50	523.2	1	474.5	4.00	182.4
H.Shoe	0.50	504.0	1	436.1	4.00	28.6
Club	0.50	521.2	1	470.5	4.00	166.2
Small	0.50	529.0	1	486.0	4.00	228.5

table 2.

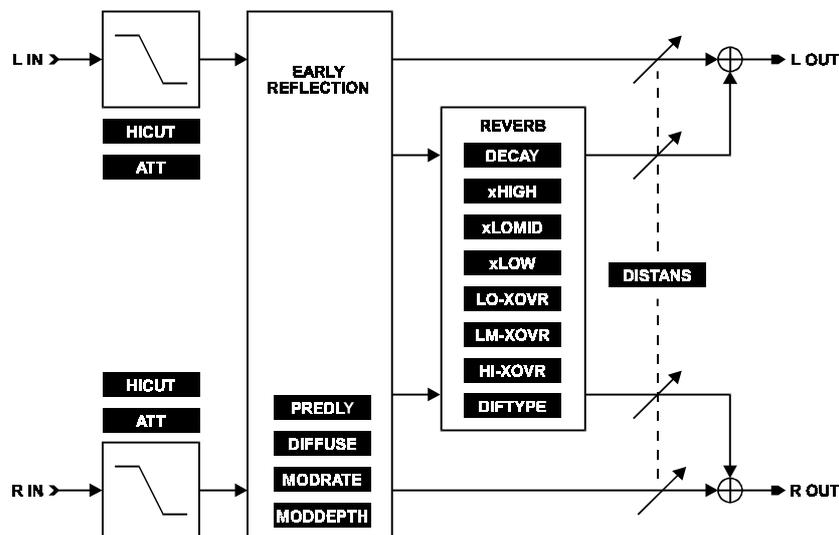
REVFEED	0.0 - 100.0 mS 0.0 - 300.0 mS ⁴	Sets the time before the reverberating part of the signal starts to build up, relative to the early reflections PREDLY.
HICUT	500 Hz - flat	High cut filter, shelving type. Provides an overall reverb high frequency rolloff (6 dB per octave) that is well suited to make a warmer sound. Sets the cut-off

⁴Only if idx RAM mounted is 64K. Check your index ram in the CONFIG menu under UTILITY.

		frequency of the overall High cut filter in 1/3-octave steps.
ATT	-40 - 0.0 dB	The attenuation control sets the high frequency rolloff determined by HICUT.
LO-XOVR	20 Hz - flat	Sets the crossover frequency for the x LOW decay time multiplier in 1/3-octave steps.
HI-XOVR	20 Hz - flat	Sets the crossover frequency for the x HIGH decay time multiplier in 1/3-octave steps.
INITLEV	off - 0.0 dB.	Sets the level of the initial pattern. The purpose of this control is to balance the initial (early) reflection levels against the reverberating part of the reverb algorithm.
REVLEV	off - 0.0 dB.	Sets the level of the reflection envelope relative to the early reflections in 0.5 dB steps. If REVLEV is set to off you will hear only the initial reflections.
RWIDTH	0 - 100 %	Sets the apparent stereo width of the reverberating part of the algorithm. At '0', the reverb tail will appear to be coming mainly from the center (mono compatible), whereas with RWIDTH set to '100' the L/R reverberators are independent.
I-XFEED	on/off	With this parameter switched off, the cross feeds in the early reflections will be killed. The I-XFEED switched off, simultaneously with the parameter RWIDTH set to 100%, will create a true stereo reverb. The effect from the left and the right channel will be generated totally independent. This is ideal for working with Dolby surround or for broadcasting in general where mono compatibility is important. The feature is also especially applicable for the film industry and post production suites.

This is a description of the parameters specific to the REVERB-3 algorithm. The REVERB-3 algorithm is very different from the REVERB-1 and 2 algorithms. It is capable of making an exceptionally clear reverb sound using a very dense and natural sounding reverb tail. DECAY time can be controlled in four individually adjustable frequency bands. Using DIFFUSE and the DISTANS (distance) control, sounds can be made in which practically no initial reflections are heard. Add to this a slight modulation to minimize room interaction with your source material and you have - REVERB-3.

The diagram below is an addition to the signal flow diagram found in the "BASIC ALGORITHMS" module, page 2.

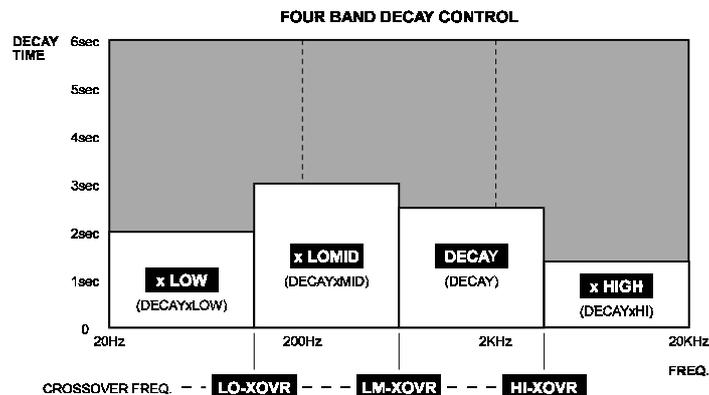


EDIT PARAMETERS:

MIX	0 - 100 %	Sets the mix between dry and wet signal. 100 % = effect signal only. Mix can be set to 100 % globally in the G-LEVELS-menu under UTILITY. Set MIXMODE WET=MAX and all dry signals are "killed" regardless of preset mix settings.
INLEV	off - 0.0 dB.	Sets the level of the input to the reverb in 0.5 dB steps. The function of the control is to maximize dynamic range. Please note that this control is positioned after the input PPM meter, and will not affect the input PPM reading, also if using an

analog input, make the analog input adjustment in the G-LEVELS-menu under UTIL before setting this control. However, if the red overload LED flashes, turn down INLEV a couple of dB's. The control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.

OUTLEV	off - 0.0 dB.	Sets the output level of the reverb in 0.5 dB steps. The function of this control is to maximize dynamic range by allowing the reverb algorithm to output maximum signal to the DA converters. It affects the output PPM reading. There is a separate output level control for adjusting the analog output level in UTIL. Set the analog input (if you use analog input) and the INLEV adjustments before setting this control. The control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.
DECAY	0.3 - 30.0 Sec.	Reverberation decay time (fig. below).
x LOW	0.01 - 2.5 times	Relative decay time multiplier for low frequencies.
x LOMID	0.01 - 2.5 times	Relative decay time multiplier for the low-mid frequencies.
x HIGH	0.01 - 2.0 times	Relative decay time multiplier for the high frequencies.

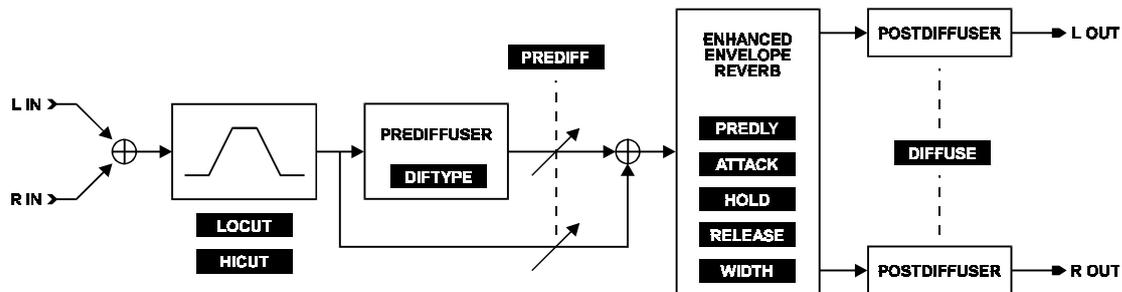


DIFFUSE	1 - 99	The DIFFUSE parameter simulates that the reflections in the room "hit" more or less uneven surfaces. With smooth walls low diffusion takes place. Walls that are uneven, with many angles, pockets or with dedicated diffusers cause the reflections to break into a high number of less identifiable reflections producing much higher diffusion. The DIFFUSE parameter affects the quality of the reverb tail as well as the spread of the initial reflections.
LO-XOVR	20 Hz - 4.00 KHz	Sets the crossover frequency for the x LOW decay time multiplier in 1/3-octave steps. If LO-XOVR is set higher than LM-XOVR then the LM-XOVR frequency will change upwards.
LM-XOVR	200 Hz - 6.30 KHz	Sets the crossover frequency for the x LOMID decay time multiplier in 1/3 octave steps. If set lower than LO-XOVR, then LO-XOVR will change downwards.
HI-XOVR	2.00 KHz - flat	Sets the crossover frequency for the x HIGH decay time multiplier in 1/3-octave steps.
PREDLY	1 - 150 mS or 1 - 470 mS ⁵	Sets the time that passes before the first reflection appear.
DISTANS	0 - 15	The relative distance control varies the mix relations between the early and the later reflections. When set to "0" more of the early reflections are heard, similar to being close to the sound source in a room. As you increase DISTANS toward "15" more of the later reflections are heard = further away from the sound source. Practically no initial reflections are heard at "15". Please note that at very short distances the initial reflections

⁵Only if idx RAM mounted is 64K. If idx=32K, max predelay will be 150 mS. M5000 automatically checks your hardware on power up and uses the available amount. The index ram size can be seen in the CONFIG menu under UTILITY.

		interact with the direct signal creating 'chorus-like' coloration's just as in real rooms with strong low-order reflections.
HICUT	500 Hz - flat	High cut filter, shelving type. Provides an overall reverb high frequency rolloff (6 dB per octave) that is well suited to make the space sound warmer.
ATT	-40 - 0.0 dB	The attenuation control sets the high frequency roll determined by HICUT.
MODRATE	1 - 200	The MODRATE varies the rate of modulation of the recirculating delay paths simulating the reverb tail. The control has no effect at a MODDPATH of "0". Adding modulation to the reverb has the effect of smoothing out the frequency response of the reverb, by effectively averaging out the room resonances.
MODDPATH	0 - 100%	Controls the amount of delay path modulation or "wander" in the reverb. The control interacts with the MODRATE, so with either control set at a high setting you will start to hear pitch modulation. The amount of either parameter that you can add depends on the type of material to which you are adding reverb. Percussive types of sounds can be much more modulated than for example violin or an opera vocal. Please note that adding even the least amount of modulation will cause the very high frequencies to diminish slightly, somewhat similar to the high frequency damping caused by sound traveling naturally through air.
DIFTYPE	Smooth1, Smooth2, Wow 1, Short1 and Short2	The natural room mode peak frequencies and the smoothness of the tail are affected by this parameter. Use Smooth1 and 2 for long decays, whereas the others are made for shorter decaytimes and to emulate the characteristics of well known plates.

This is a brief description of the NONLIN-1 parameters. With the NONLIN-1 algorithm a number of gated reverb type sounds and non-linear rooms can be created. By non-linear rooms we mean reverb sounds that cannot be made by any real room equivalent. A non-linear example typically has a fast build-up and sudden decay reverb, very useful for drum work. Another is that of a 'reverse room' by making a gradual build-up and sudden decay. The NONLIN-1 algorithm features 3 powerful controls for shaping the dynamics of the reverb pattern: ATTACK, HOLD and RELEASE as well as selection of the underlying reflection pattern; DIFTYPE, density control; DIFFUSE, plus stereo width and color controls. Please note that unlike a reverb plus a gate/expander device, this algorithm is completely level and time independent, i.e. each drumbeat gets identical and independent reverb 'tails' added, regardless of the level or how fast the beats are played in succession. The 'secret' behind this is the powerful M5000 initial pattern capabilities. The basic effect is produced by a very long, shapeable non-recirculating pattern of reflections. The diagram below is an addition to the signal flow diagram found in the "BASIC ALGORITHMS" module, page 2.



EDIT PARAMETERS:

PAGE 1:

MIX	0 - 100 %	Sets the mix between dry and wet signal. 100 % = effect signal only. Mix can be set to 100 % globally in the G-LEVELS-menu under UTILITY. Set MIXMODE WET=MAX and all dry signals are "killed" regardless of preset mix settings.
INLEV	off - 0.0 dB	Sets the level of the input to the program in 0.5 dB steps. The function of the control is to maximize dynamic range. Please note that this control is positioned after the input PPM meter, and will not affect the input PPM reading, also if using an analog input, make the analog

input adjustment in the G-LEVELS-menu under UTIL before setting this control. If the red overload LED still flashes, turn down INLEV a couple of dBs. The control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.

OUTLEV off - 0.0 dB

Sets the output level of the program in 0.5 dB steps. The function of this control is to maximize dynamic range by allowing the chorus algorithm to output maximum signal to the DA converters. It affects the output PPM reading. There is a separate output level control for adjusting the analog output level in UTIL. Set the analog input (if you use analog input) and the INLEV adjustments before setting this control. The control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.

PAGE 2:

PREDLY 0 - 490 mS*

Sets the time that passes before the first reflection of the initial pattern appears.

ATTACK 0 - 490 mS*

Determines the time of the attack part in the non-linear reflection envelope.

HOLD 10 - 500 mS*

Determines the time that the gate reverb is open until RELEASE starts to decay.

RELEASE 0 - 490 mS*

Determines the time of the decay in the non-linear reflection envelope.

* As the total non-linear reflection pattern has a fixed length, the maximum time of the above parameters will depend on each others settings whose total cannot exceed 500 mS with standard memory.

PAGE 3:

LOCUT 20 - 2.00 KHz

Low cut filter, shelving type. Provides an overall low frequency rolloff (6 dB per octave). Sets the cutoff frequency of the overall low cut filter in 1/3-octave steps.

HICUT 800 - flat

High cut filter, shelving type. Provides an overall high frequency rolloff (6 dB

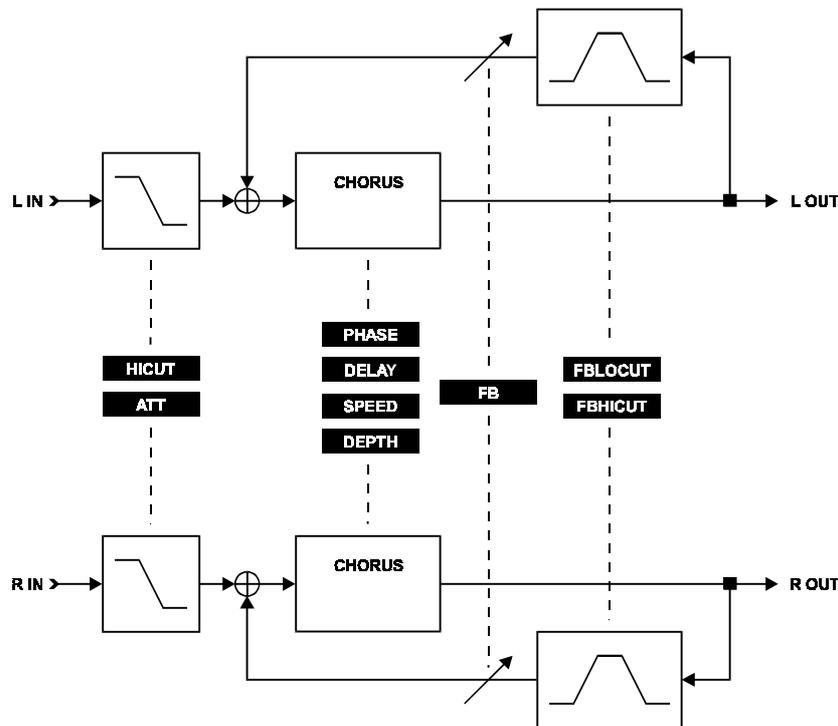
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per octave) Sets the cutoff frequency of the overall high cut filter in 1/3-octave steps.

PAGE 4:

DIFFUSE	0 - 25	Simulates the reflections in the room "hit" a more or less uneven surface. The DIFFUSE parameter affects the density of the gated reverb. To set the DIFFUSE properly, adjust while listening on percussive type of signals or instruments. High DIFFUSE settings might add some release time.
PREDIFF	0 - 100	Adds extra diffusion to the non-linear reverb. PREDIFF is a mix function which adds prediffusion from the selected DIFTYPE.
DIFTYPE	BRIGHT1, BRIGHT2, WARM, MIDTONE	The patterns used for prediffusion. The 4 types have different 'color'-characteristics. The prediffusion is mixed into the reverb by PREDIFF.
WIDTH	0 - 100 %	Sets the apparent stereo width of the algorithm. At '0' the gated reverb will appear to be coming mainly from the center (mono compatible), whereas with WIDTH set to '100' the L/R reverberators are independent.

The following is a brief description of the CHORUS-1 algorithm. This algorithm produces normal chorus, flanging and to some extent, delay-effects, digitally. The algorithm is also capable of overdoing the effect in order to create some "wild" sounds. The diagram below is an addition to the signal flow diagram found in the "BASIC ALGORITHMS" module, page 2.



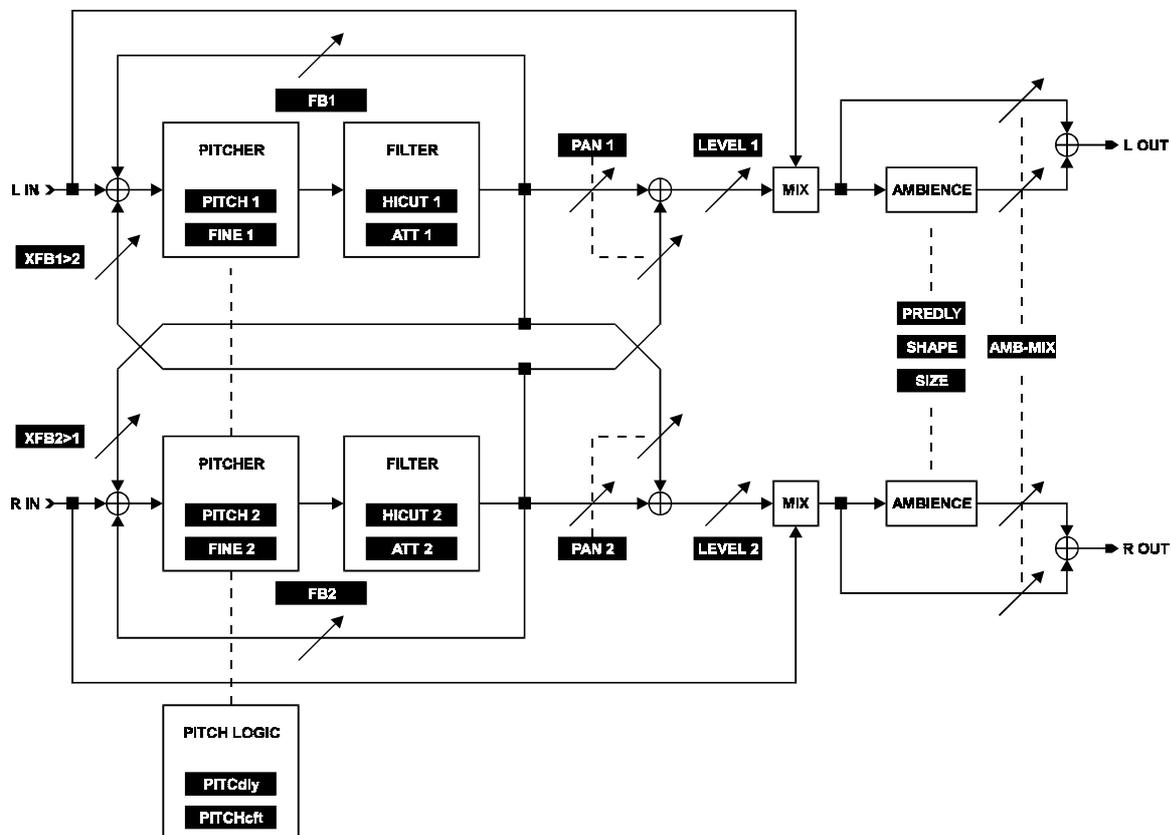
EDIT PARAMETERS:

MIX	0 - 100 %	Sets the mix between dry and wet signal. 100 % = effect signal only. Mix can be set to 100 % globally in the G-LEVELS-menu under UTILITY. Set MIXMODE WET=MAX and all dry signals are "killed" regardless of preset mix settings.
INLEV	off - 0.0 dB.	Sets the level of the input to the program in 0.5 dB steps. The function of the control is to maximize dynamic range. Please note that this control is positioned after the input PPM meter, and will not affect the input PPM reading, also if using an analog input, make the analog input adjustment in the G-LEVELS-

		<p>menu under UTIL before setting this control. If the red overload LED still flashes, turn down INLEV a couple of dBs. The control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.</p>
OUTLEV	off - 0.0 dB.	<p>Sets the output level of the program in 0.5 dB steps. The function of this control is to maximize dynamic range by allowing the chorus algorithm to output maximum signal to the D/A converters. It affects the output PPM reading. There is a separate output level control for adjusting the analog output level in UTIL. Set the analog input (if you use analog input) and the INLEV adjustments before setting OUTLEV. This control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.</p>
PHASE	0° - 90° - 180°	<p>Determines the sine wave modulation phase shift between left and right channels. At 0° the left and right modulation will move in sync. At 180° the modulation will move the channels against each other.</p>
DELAY	0 - 670 mS (Idx=32K) 0 - 1.360 mS (Idx=64K)	<p>Controls the length of delay time. Max delaytime. Depends on the index RAM in the machine (also called high memory). Check how much index RAM you have in the utility menu CONFIG.</p>
FEEDBACK	0 - 99 %	<p>Controls the amount of effect signal routed back to the chorus input (Flanging).</p>
SPEED	0.1 Hz - 10 Hz	<p>Controls the rate of sweep in a range from 1 sweep every 10 seconds to 10 sweeps every second.</p>
DEPTH	0 - 100 %	<p>Determines how wide a modulation (sweep) is produced.</p>

FBLOCUT	off - 800 Hz	Feedback Low-Cut enables you to remove low frequencies from the feedback loop.
FBHICUT	1 KHz - off	Feedback High-Cut enables you to remove high frequencies from the feedback loop.
HICUT	500 Hz - flat	High-cut filter enables you to make the chorus sound more "warm". This is a 6 dB per octave filter.
ATT	-40 - 0.0 dB	Gain for HICUT filter. Adjustable in 0.5 dB steps.

One of the common purposes for using a pitch shifter is to get the instrument or vocalist to sound "richer" as a plain effect. Yet, through time the pitch shifter has become more intelligent and the purposes more complicated. Today there are several different forms of pitch shifters which can be used in many different applications. An instrument or maybe more obvious - a vocalist who sings a bit out of tune can through the use of a pitch shifter appear to sing in key. Another use is to produce harmonies with a single source signal, creating your own choir in real time. In the basic software there are a few high quality pitch shifting algorithms that demonstrates the power of the M5000. Specific for the REVPITCH algorithm you are able to add some ambience to the signal. The diagram below is an addition to the signal flow diagram found in the "BASIC ALGORITHMS" module, page 2.



A pitch shift effect is produced as the source signal is replayed either faster (pitch up) or slower (pitch down) The signals can then be mixed and the harmonies will be produced.

PITCH UP

In order to replay the signal faster, some chosen "parts" have to be repeated simultaneously with the original signal. This is called LOOP BACK. The selection of these parts are of vital importance for the quality of the pitch and are completely controlled by the software.

PITCH DOWN

This is the opposite situation where chosen parts of the signal must be skipped. This is called LOOP FORWARD. Again, the selection of the parts are essential for the quality of the pitch. To avoid major disturbances caused by the repeating/skipping of parts in the signal, the distance of the inserted or removed parts must be as short as possible.

EDIT PARAMETERS:

PAGE 1:

MIX	0 - 100 %	Sets the mix between dry and wet signal. 100 % = effect signal only. Mix can be set to 100 % globally in the G-LEVELS-menu under UTILITY. Set MIXMODE WET=MAX and all dry signals are "killed" regardless of preset mix settings.
INLEV	off - 0.0 dB	Sets the level of the input to the program in 0.5 dB steps. The function of the control is to maximize dynamic range. Please note that this control is positioned after the input PPM meter, and will not affect the input PPM reading, also if using an analog input, make the analog input adjustment in the G-LEVELS-menu under However, if the red overload LED flashes, turn down INLEV a couple of dBs. The control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.
OUTLEV	off - 0.0 dB	Sets the output level of the program in 0.5 dB steps. The function of this control is to maximize dynamic range by allowing the program to output maximum signal to the DA converters. It affects the output PPM reading. Note that there is a separate output level control for adjusting the analog output

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level. Set the analog input (if you use analog input) and the INLEV adjustments before setting this control. The control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.

PAGE 2:

PITCH 1	-12 - 12	Transposition for pitch shifter 1. One step corresponds to a semitone (one half-step). 0 corresponds to no pitch shift and 12 or -12 is equal to one octave up or one octave down.
FINE 1	-50 - 50	Fine adjustment of pitch shifter 1. When set to 0 there is no fine adjustment. -50 or 50 is equal to one semitone down or up.
PITCH 2	-12 - 12	Transposition for pitch shifter 2. One step corresponds to a semitone (one half-step). 0 corresponds to no pitch shift and 12 or -12 is equal to one octave up or one octave down.
FINE 2	-50 - 50	Fine adjustment of pitch shifter 2. When set to 0 there is no fine adjustment. -50 or 50 is equal to one semitone down or up.

PAGE 3:

LEVEL 1	off - 0.0 dB	In order to match the balance between the 2 pitches or/and the original (dry) signal LEVEL 1 sets the level on PITCH 1 only.
PAN 1	50L - center - 50R	PAN separates the pitches between left and right. When PAN 1 is set to "50L" the PITCH 1 will appear in the left side.
LEVEL 2	off - 0.0 dB	Like LEVEL 1, LEVEL 2 sets the level on PITCH 2 instead.
PAN 2	50L - center - 50R	When PAN 2 is set to "50R" the PITCH 2 will appear in the right side.

PAGE 4:

HICUT 1	500 Hz - flat	High cut filter, shelving type for PITCH 1. Provides an overall high frequency
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rolloff (6 dB per octave) that is well suited to make the pitch more warm sounding. Sets the cutoff frequency of the overall high cut filter in 1/3-octave steps.

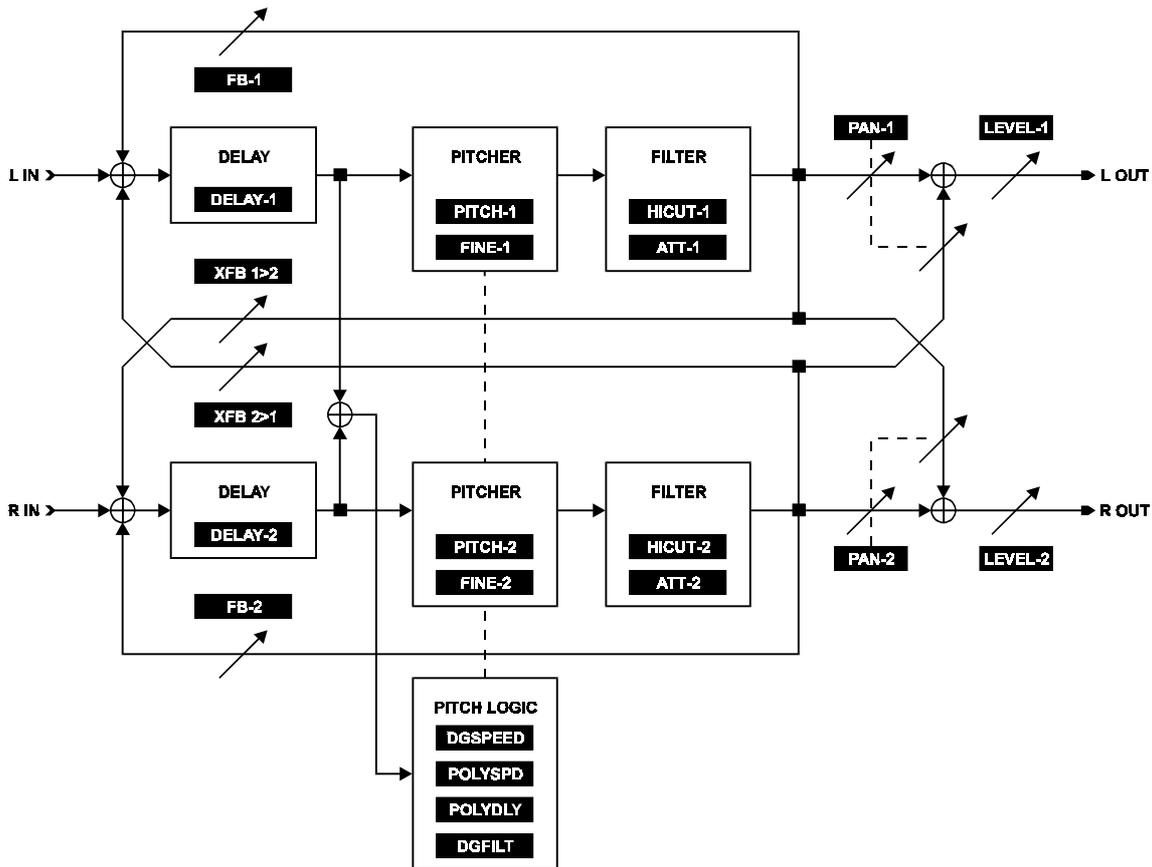
ATT 1	-40 dB - 0.0 dB	The attenuation control sets the high frequency rolloff determined by HICUT 1 in 0.5 dB steps.
HICUT 2	500 Hz - flat	High cut filter, shelving type for PITCH 2. Provides an overall high frequency rolloff in 6 dB per octave.
ATT 2	-40 dB - 0.0 dB	The attenuation control sets the high frequency roll off determined by HICUT 2 in 0.5 dB steps.

PAGE 5:

FB 1	0 -100	Feedback for PITCH 1. Returns the pitch output to its own input. This is for creating a more powerful and fat sounding effect. Set the FB 1 > "0" to get pitch smears. The more FB the more powerful effect.
FB 2	0 - 100	Same as FB 1.
XFB 1>2	0 - 100	Cross feedback. Returns the PITCH 1's output to PITCH 2's input. With this feature you are able to create some wild effects. If PITCH 1 is pitching down the effect can be pitched even lower by routing it to the PITCH 2's input which then must be set to pitch down.
XFB 2>1	0 - 100	Same as XFB 1>2 only it works vice versa.
AMB-MIX	0 - 100 %	Mix level of the amount of ambiance/reverb added to the pitch effect.
PREDLY	0 - 150 mS	Sets the time that passes before the initial reflection pattern starts.
SHAPE	HALL, FAN, PRISM, H.SHOE, CLUB, SMALL.	Initial reflection pattern. The different room-shapes has different characteristics. Please refer to the REVERB-1 & 2 algorithm text module

	DELAY	for further description of the different shapes. This is only one reflection (tab). With this shape it will act as a normal digital delay.
x SIZE	0.040 - 4.000	Scales the dimensions of the simulated space depending on the SHAPE chosen. A detailed description can be found under the REVERB-1 algorithm text module.
PAGE 7:		
PITCdly	10 - 40 mS	Maximum pitch transition delay. The more delay the better quality of the pitch.
PITCcft	5 - 100	Per cent of the pitch delay used for crossfade. Must be tuned in order to minimize the tremolo effect. Best setting depends on the type of input signal.

The PITCH-1 algorithm is an ultra-fast and high resolution harmony effect with an intelligent working de-glitcher. The algorithm has two pitch shifters each of which can be panned in the stereo image. The pitch shifters have independent pitch, filter, feedback and delay settings. It is also possible to crossfeed from one pitcher to the other whereby existing pitch harmony build-ups can be made. The diagram below is an addition to the signal flow diagram found in the "BASIC ALGORITHMS" module, page 2.



EDIT PARAMETERS:

PAGE 1:

MIX	0 - 100 %	Sets the mix between dry and wet signal. 100 % = effect signal only. Mix can be set to 100 % globally in the G-LEVELS-menu under UTILITY. Set MIXMODE WET=MAX and all dry signals are "killed" regardless of preset mix settings.
INLEV	off - 0.0 dB.	Sets the level of the input to the program in 0.5 dB steps. The function of the

control is to maximize dynamic range. Please note that this control is positioned after the input PPM meter, and will not affect the input PPM reading, also if using an analog input, make the analog input adjustment in the G-LEVELS-menu under UTIL before setting this control. If the red overload LED still flashes, turn down INLEV a couple of dBs. The control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.

OUTLEV off - 0.0 dB.

Sets the output level of the program in 0.5 dB steps. The function of this control is to maximize dynamic range by allowing the pitch algorithm to output maximum signal to the DA converters. It affects the output PPM reading. There is a separate output level control for adjusting the analog output level in UTIL. Set the analog input (if you use analog input) and the INLEV adjustments before setting OUTLEV. This control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.

PAGE 2:

PITCH-1	-12 - +12	Pitch shift for channel 1 (in semitones).
FINE-1	-1200 - +1200	Pitch shift for channel 1 (in cents).
PITCH-2	-12 - +12	Pitch shift for channel 2 (in semitones).
FINE-2	-1200 - +1200	Pitch shift for channel 2 (in cents).

PAGE 3:

LEVEL-1	off - 0.0 dB	The output level of channel 1.
PAN-1	50L - center - 50R	Controls the position of channel 1 in the stereo image.
LEVEL-2	off - 0.0 dB	The output level of channel 2.
PAN-2	50L - center - 50R	Controls the position of channel 2 in the stereo image.

PAGE 4:

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HICUT-1	500 Hz - flat	High cut filter for channel 1. Enables you to make the pitch-transposer more "warm". This is a 6 dB per octave filter.
ATT-1	-40 - 0.0 dB	Gain for HICUT filter. Adjusts in 0.5 dB steps.
HICUT-2	500 Hz - flat	High cut filter for channel 2 Enables you to make the pitch-transposer more "warm". This is a 6 dB per octave filter.
ATT-2	-40 - 0.0 dB	Gain for HICUT filter. Adjusts in 0.5 dB steps.
PAGE 5:		
FB-1	0 - 100	The percentage of feedback for channel 1 (feedback path includes delay, pitch and hi-cut).
FB-2	0 - 100	The percentage of feedback for channel 2 (feedback path includes delay, pitch and hi-cut).
XFB 1>2	0 - 100	The percentage of crossfeed from channel 1's output to ch. 2's input.
XFB 2>1	0 - 100	The percentage of crossfeed from channel 2's output to ch. 1's input.
PAGE 6:		
DELAY-1	0 - 310 mS	The delay setting for channel 1.
DELAY-2	0 - 310 mS	The delay setting for channel 2.
PAGE 7:		
DGSPEED	0.05 - 0.5	The deglitch speed parameter should be set relatively low for slowly changing and monophonic source material. Higher settings are for fast changing and polyphonic material.
POLYSPD	5 - 50	The polyphonic speed parameter should be set high for polyphonic and bass type sources.
POLYDLY	5 - 18	The polyphonic delay parameter controls the response to polyphonic signals. When this parameter is turned up the response time will be slower, but the

DGFILT

500 Hz, 1 kHz, 2 kHz,
and 4 kHz.

ability to de-glitch polyphonic chords
will be enhanced.

This filter is used to determine the upper limit of frequencies of your input signal. The idea is to make the frequency range, within the pitch shifter, narrower. This will increase the speed of the pitch shifting, because the pitch detector doesn't have to search for so many frequencies in order to determine the pitch of the input signal. This conclusion is maybe more understandable if we use the following analogy: Let's say you have lost your car keys. It is quite likely that you would be able to locate them faster, if you knew they were somewhere in your garage, than if you knew they were somewhere in your house.

PAGE 5:

MIDIPTc

off, on

MIDI Pitch Bender control. If this is set to 'on' the PITCH-1 algorithm will react on Pitch Bender control from a MIDI keyboard.

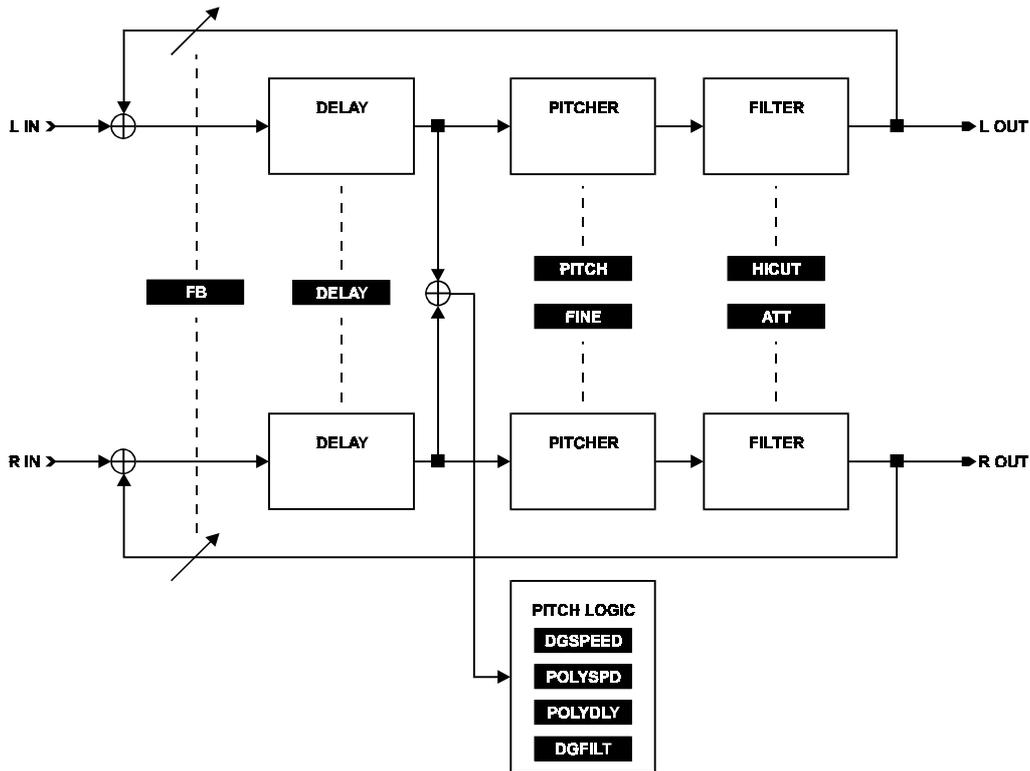
MIN

-1200 - 1200

This determines the minimum key range value for the Pitch Bender Wheel.

MAX -1200 - 1200 This determines the maximum key range value for the Pitch Bender Wheel.

The PITCH-2 algorithm is an ultra-fast and high resolution harmony effect with an intelligent working de-glitcher. The difference from the PITCH-1 algorithm is, that this is a stereo pitch-transposer where the left and the right channels are linked together to ensure a 100% phase linear output. The diagram below is an addition to the signal flow diagram found in the "BASIC ALGORITHMS" module, page 2.



EDIT PARAMETERS:

PAGE 1:

MIX	0 - 100 %	Sets the mix between dry and wet signal. 100 % = effect signal only. Mix can be set to 100 % globally in the G-LEVELS-menu under UTILITY. Set MIXMODE WET=MAX and all dry signals are "killed" regardless of preset mix settings.
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INLEV	off - 0.0 dB.	<p>Sets the level of the input to the program in 0.5 dB steps. The function of the control is to maximize dynamic range. Please note that this control is positioned after the input PPM meter, and will not affect the input PPM reading, also if using an analog input, make the analog input adjustment in the G-LEVELS-menu under UTIL before setting this control. If the red overload LED still flashes, turn down INLEV a couple of dBs. The control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.</p>
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OUTLEV	off - 0.0 dB.	<p>Sets the output level of the program in 0.5 dB steps. The function of this control is to maximize dynamic range by allowing the pitch algorithm to output maximum signal to the DA converters. It affects the output PPM reading. There is a separate output level control for adjusting the analog output level in UTIL. Set the analog input (if you use analog input) and the INLEV adjustments before setting OUTLEV. The control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.</p>
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PAGE 2:

PITCH	-12 - +12	Pitch shifting in semitones.
FINE	-1200 - +1200	Pitch shifting in cents.
FB	0 - 100%	The percentage of feedback. (Feedback path includes delay, pitch and hicut).
DELAY	0 - 310 mS	The delay setting. Sets the delay before the signal is pitched.

PAGE 3:

HICUT	500 Hz - flat	High cut filter enables you to make the pitched signal more "warm". This is a 6 dB per octave filter.
ATT	-40 - 0.0 dB	Gain for HICUT filter. Adjusts in 0.5 dB steps.

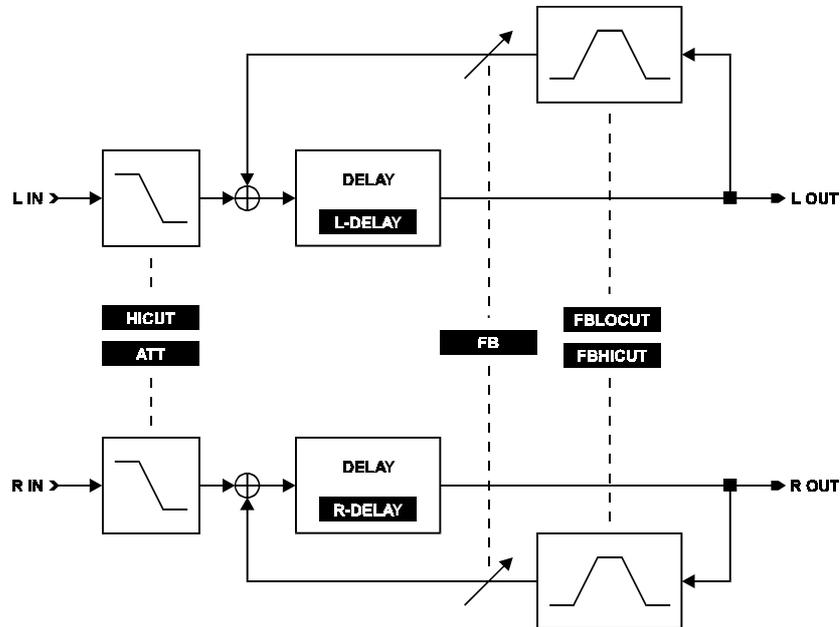
PAGE 4:

DGSPEED	0.05 - 0.5	The de-glitch speed parameter should be set relatively low for slowly changing and monophonic source material. Higher settings are for fast changing and polyphonic material.
POLYSPD	5 - 50	The polyphonic speed parameter should be set high for polyphonic and bass type sources.
POLYDLY	5 - 18	The polyphonic delay parameter control the response to polyphonic signals. When this parameter is turned up the response time will be slower, but the ability to deglitch polyphonic chords will be enhanced.
DGFILT	500 Hz, 1 KHz, 2 KHz, and 4 KHz	This filter is used to determine the upper limit of frequencies in your input signal. The idea is to make the frequency range, within the pitch shifter, narrower. This will increase the speed of the pitch shifting, because the pitch detector doesn't have to search for so many frequencies to determine the pitch of the input signal.

PAGE 5:

MIDIptc	off, on	MIDI Pitch Bender control. If this is set to 'on' the PITCH-2 algorithm will react on Pitch Bender control from a MIDI keyboard.
MIN	-1200 - 1200	This determines the minimum key range value for the Pitch Bender Wheel.
MAX	-1200 - 1200	This determines the maximum key range value for the Pitch Bender Wheel.

The DELAY-1 algorithm is basically a simple and easy to handle true stereo digital delay line. The diagram below is an addition to the signal flow diagram found in the "BASIC ALGORITHMS"-module, page 2.



EDIT PARAMETERS

PAGE 1:

MIX	0 - 100 %	Sets the mix between dry and wet signal. 100 % = effect signal only. Mix can be set to 100 % globally in the G-LEVELS-menu under UTILITY. Set MIXMODE WET=MAX and all dry signals are "killed" regardless of preset mix settings.
INLEV	off - 0.0 dB.	Sets the level of the input to the program in 0.5 dB steps. The function of the control is to maximize dynamic range. Please note that this control is positioned after the input PPM meter, and will not affect the input PPM reading, also if using an analog input, make the analog INLEV adjustment in the G-LEVELS-menu under UTIL before setting this control. If the red overload LED flashes, turn down INLEV a couple of dBs. The control does not affect the bypassed

OUTLEV	off - 0.0 dB.	<p>signal level. Normally, you do not have to change the factory default setting.</p> <p>Sets the output level of the program in 0.5 dB steps. The function of this control is to maximize dynamic range by allowing the delay algorithm to output maximum signal to the DA converters. It affects the output PPM reading. Note that there is a separate output level control for adjusting the analog output level. Set the analog OUTLEV adjustment in the G-LEVELS-menu before setting this control. This control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.</p>
 PAGE 2:		
L-DELAY	1 - 670 mS (1.36 Sec.) ⁶	Sets the delay time for the left side.
R-DELAY	1 - 670 mS (1.36 Sec.) ¹	Sets the delay time for the right side.
FB	0 - 99 %	Sets common feedback level for left and right delay output in percent. It feeds the delay output for left and right separately to its own input in order to make repeatable stereo echo effects. The control is common for left and right - but the signals are processed individually.
 PAGE 3:		
FBLOCUT	off - 800 Hz	Common low cut filter control for left and right feedback.
FBHICUT	1 KHz - off	Common high cut filter control for left and right feedback.
HICUT	500 Hz - flat	High cut filter, shelving type. Provides an overall high frequency rolloff (6 dB per octave) that is well suited to make the delay effect sound warmer Sets the

⁶If high memory is installed (Idx=64K check your index memory under UTILITY menu CONFIG) max. delay time is 1.36 Sec., otherwise max. delay time is 670 mS. (Idx=32K).

ATT

-40 - 0.0 dB

cutoff frequency of the overall high cut filter in 1/3-octave steps.

The attenuation control sets the high frequency roll determined by HICUT in 0.5 dB steps.

The DELAY-2 algorithm is an advanced but easy to handle true stereo digital delay line. With cross feedback section and modulation section, this delay algorithm is capable of doing anything from smooth spatial expanding to the wildest echo effects. The diagram below is an addition to the signal flow diagram found in the "BASIC ALGORITHMS"-module, page 2.

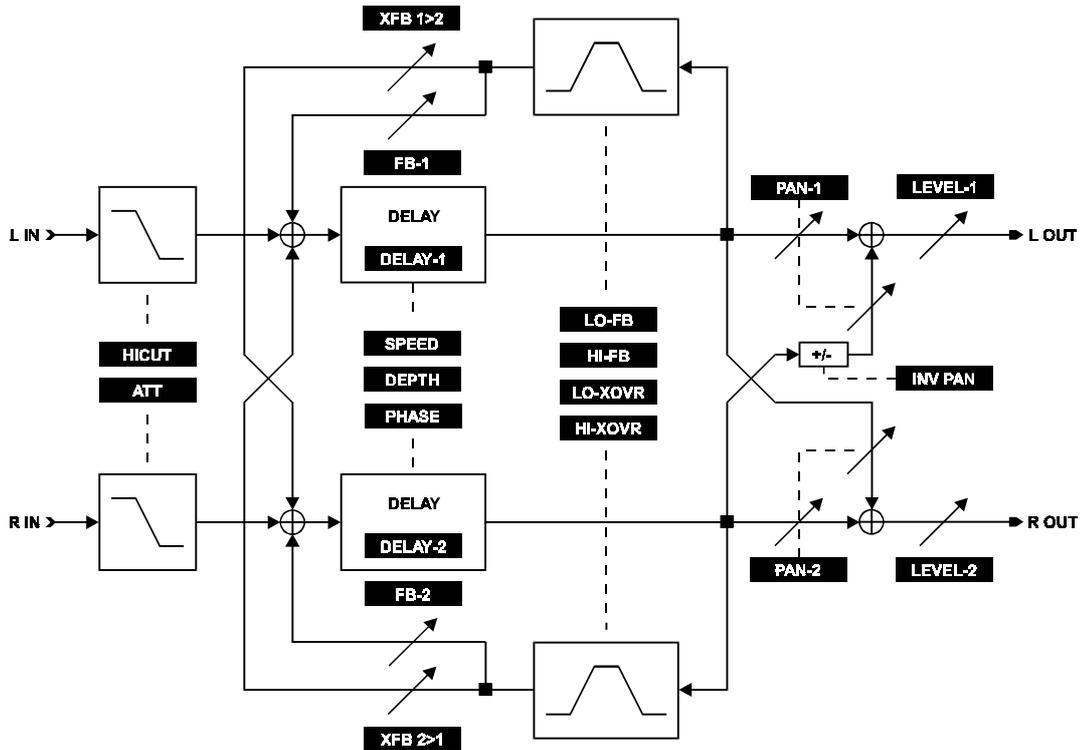


fig. 1

EDIT PARAMETERS

PAGE 1:

MIX	0 - 100 %	Sets the mix between dry and wet signal. 100 % = effect signal only. Mix can be set to 100 % globally in the G-LEVELS-menu under UTILITY. Set MIXMODE WET=MAX and all dry signals are "killed" regardless of preset mix settings.
INLEV	off - 0.0 dB.	Sets the level of the input to the program in 0.5 dB steps. The function of the control is to maximize dynamic range. Please note that this control is positioned

after the input PPM meter, and will not affect the input PPM reading, also if using an analog input, make the analog INLEV adjustment in the G-LEVELS-menu under UTIL before setting this control. If the red overload LED flashes, turn down INLEV a couple of dBs. The control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.

OUTLEV off - 0.0 dB.

Sets the output level of the program in 0.5 dB steps. The function of this control is to maximize dynamic range by allowing the delay algorithm to output maximum signal to the DA converters. It affects the output PPM reading. Note that there is a separate output level control for adjusting the analog output level. Set the **analog** OUTLEV adjustment in the G-LEVELS-menu before setting this OUTLEV. The control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.

PAGE 2:

DELAY-1 1 - 670 mS (1.36 Sec.)⁷

Sets the delay time for the left side.

DELAY-2 1 - 670 mS (1.36 Sec.)¹

Sets the delay time for the right side.

HICUT 500 Hz - flat

High cut filter, shelving type. Provides an overall high frequency rolloff (6 dB per octave) that is well suited to make the delay effect sound warmer. Sets the cutoff frequency of the overall high cut filter in 1/3-octave steps.

ATT -40 - 0.0 dB

The attenuation control sets the amount of high frequency rolloff determined by HICUT in 0.5 dB steps.

⁷If high memory is installed (Idx=64K check your index memory under UTILITY menu CONFIG) max. delay time is 1.36 Sec., otherwise max. delay time is 670 mS. (Idx=32K).

PAGE 3:

LEVEL-1	off - 0.0 dB	The output level of channel 1.
PAN-1	50L - center - 50R	Controls the position of channel 1 in the stereo image.
LEVEL-2	off - 0.0 dB	The output level of channel 2.
PAN-2	50L - center - 50R	Controls the position of channel 2 in the stereo image.

PAGE 4:

SPEED	0.1 Hz - 10 Hz	Controls the rate of modulation sweeps in a range from 1 sweep every 10 seconds to 10 sweeps per second.
DEPTH	0 - 100%	Determines how wide a modulation sweep is produced. If you do not want to modulate the effect signal, set this parameter to 0%.
PHASE	0° - 90° - 180°	Determines the sine wave modulation phase shift between left and right channel. At 0° the left and right channel will move in sync. At 180° the modulation will move against each other.
INV PAN	on/off	Inverts the phase of ch.2 effect signal panned to ch.1 output (see fig.1). With INV PAN "on" it is possible to make sum/difference type outputs that work well for spatial (TC 1210 alike) effects.

PAGE 5:

The numeric sum value of the feedback- and crossfeed parameters may not exceed 200. Any value above 200 may cause oscillation.

FB-1	-100 - 100%	The percent of positive phase and negative phase feedback for channel 1 (feedback path includes low-cut and hi-cut filters on PAGE 6).
FB-2	-100 - 100%	The percent of positive phase and negative phase feedback for channel 2 (feedback path includes low-cut and hi-cut filters on PAGE 6).
XFB 1>2	-100 - 100%	The percent of crossfeed from channel 1 output to ch. 2 input.

XFB 2>1 -100 - 100% The percent of crossfeed from channel 2 output to ch. 1 input.

PAGE 6:

LO-FB -40.0 dB - 0.0 dB Gain for LO-XOVR filter.
Adjust in 0.5 dB.

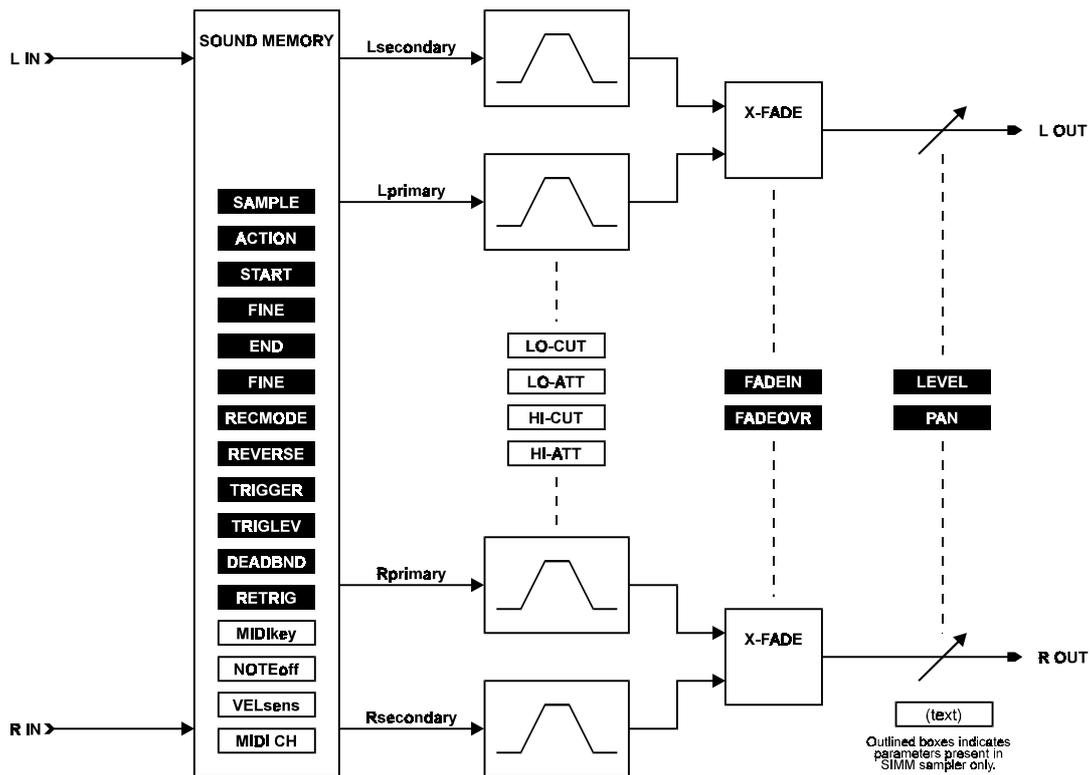
HI-FB -40.0 dB - 0.0 dB Gain for HI-XOVR filter.
Adjusts in 0.5 dB.

LO-XOVR 20 Hz - flat Frequency for 6 dB pr. octave low-cut filter.

HI-XOVR 20 Hz - flat Frequency for 6 dB pr. octave hi-cut filter.

This algorithm is very similar to our popular sample option in the TC 2290. Some of the main differences are however, that this sampler features **STEREO** sampling, the sample can be played with MIDI velocity and the samples can be loaded and saved to disk. Several samples can be stacked in the internal memory and all samples can be replayed simultaneously from a MIDI keyboard/sequencer, e.g. a drum machine with sampled drum sounds. Another common use is "flying in" vocal samples. Let's say you need backing vocals on your song. For this purpose you hire one or more vocalists. Normally, they would have to sing the same chorus lines several times during the song and maybe it needs to be overdubbed with harmonies. All this takes time which in the end means money. With a sample flyer you have the backing singers sing several chorus versions on for example 10-15 different tracks. This might take only a few hours. Once the backing singers have left, the engineer/producer is able to easily arrange and mix a complete version of a chorus. When this is done the complete mix of a chorus is sampled into the M5000 in stereo. Now he can 'fly' the chorus into the song wherever it's needed. The chorus can of course be saved to disk for later remixing.

The diagram below is an addition to the signal flow diagram found in the "BASIC ALGORITHMS" module on page 2.



STANDARD OR SIMM SAMPLER:

With software version 1.15 the SAMPLE-1 is implemented in a full featured version. All you have to do is to install some SIMM memory modules (CONFIGURATION Section, SIMM INSTALLATION module). When no SIMM memory is installed the so called STANDARD SAMPLER will be active and some parameters may be adjusted but has no effect. As soon SIMM is installed the SIMM SAMPLER will be active and all parameters will be fully available. The inactive parameters in the STANDARD SAMPLER will in the following be marked with (*).

PROGRAM PARAMETERS*:

When storing a sample preset no sound is actually saved, only a number of setup parameters for use when recording and playing back samples. A SAMPLE-1 preset holds the following parameters:

MIX, INLEV, OUTLEV, RECMODE, FIL-RES, FADEIN, FADEOVR, TRIGGER, TRIGLEV, DEADBND and RETRIG.

Note that recalling another preset merely changes the current setting of these parameters.

SAMPLE PARAMETERS:

The following parameters are attached to each sample and are also kept with the individual sample when saving to/loading from disk.

RECMODE, REVERSE, LEVEL, PAN, filter settings, MIDIkey, NOTEoff and VELsens.

EDIT PARAMETERS:

MIX	0 - 100 %	Sets the mix between direct and sampled signal. In order to monitor the input signal, press the bypass button. MIX can be set to 100% globally in the G-LEVELS-menu under UTILITY. Set MIXMODE WET=MAX and all direct signals are "killed" regardless of preset mix settings.
INLEV	off - 0.0 dB.	Sets the level of the input to the sampler in 0.5 dB steps. The function of the control is to maximize dynamic range. Please note that this control is positioned after the input PPM meter, and will not affect the input PPM reading. Also when using an analog input, make the analog input adjustment in the G-LEVELS-menu under UTIL before setting this

OUTLEV	off - 0.0 dB.	control. However, if the red overload LED flashes, turn down INLEV a couple of dBs. This control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.
		Sets the output level of the sampler in 0.5 dB steps. The function of this control is to maximize dynamic range by allowing the sampler to output maximum signal to the DA converters. It affects the output PPM reading. There is a separate output level control for adjusting the analog output level in UTIL. Set the analog input (if you are using analog input) and the INLEV adjustments before setting this control. The control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.
MEMORY	std., simm	Shows whether you are running the standard or SIMM sampler version.
PAGE 2:		
SAMPLE*	Sample selector	Shows current sample. When a sample has been made the text 'new..' is renamed to ' sampl(xx) '. If you load a sample from disk, the name of the sample is displayed here. Several samples can be stacked within the limit of available memory (see FREEMEM). You select a sample by turning softdial A.
ACTION	none play	This parameter lets you decide your actions with the sampler: This is a "safe" mode. No action taken. This is for playback of selected sample. Play the sample by pressing the DO-button. You can manually re-trigger the sample by pressing the DO-button.

play tr	Select this mode in order to enable audio triggering. Manual start of sample can still be done from the DO-button. When the FAST TRIG chip is installed on the AD/DA converter board, analog audio triggering can especially be used for fast drum triggering. Be aware that the FAST TRIG* is an 8 bit converter device, so do not set the MIX to anything other than 100%. You can hear the low quality (but fast) 8 bit trigger signal if you bypass when armed for audio triggering.
loop	After pressing the DO-button the selected sample will loop the sample in its full length from start point to end point. You can abandon the loop by pressing the UNDO-button. Everytime the DO-button is pressed the sample is started from the start point.
rec.	Start recording by pressing DO and stop recording by pressing UNDO. If UNDO is not pressed the recording will continue to the end of total sample time (FREE-MEM). You can then edit your sample as you wish.
rec. tr	This mode enables the possibility to start recording from the audio inputs. Manual start with DO-button is also possible.
delete*	This is for deleting one of the samples. Select the sample you want to delete and press DO and confirm the deletion. CAUTION: When a sample is deleted you can <u>not</u> undo the event ! After deleting one of the samples, the rest automatically are packed in memory so maximum sampletime again is available.
pack*	When a sample has been truncated using the start and end point parameters (see page 3) the action can be made permanent and the available memory is "packed" in order to free deleted space for new samples. CAUTION: You can <u>not</u> undo a "pack" event, the sample is trimmed permanently !

load*

Loads a sample from floppy disk. When a disk is inserted and DO is pressed the program dial scrolls through the samples on disk. Press DO to load the selected sample. While loading the sample, the sample rate, the FIL-RES used when saved, and whether it is mono or stereo, is shown in the display.

save*

This mode stores the selected sample on floppy disk. When DO is pressed you can change the sample filename. Press DO again and the sample is stored on disk. The filesize is depended on the word-width specified with FILE-RES.

The *.wav file is in a RIFF format, (As specified in Windows SDK multimedia file format) which means that it holds a little 'WAVE' header with information on data format (PCM), number of channels, (mono/stereo), sampling rate, bufferinfo, block align info, and a tc 'chunk' with information on: reverse, level, pan, filter settings, MIDI key note & velocity sense flag.

MAX. SAMPLE TIME PER FLOPPY DISK					
Samplerate	Wordsize bits	Disk size 1.44M		Disk size 720K	
		Stereo	Mono	Stereo	Mono
@ 48.0KHz	24	5,0 Sec.	10,0 Sec.	2,5 Sec.	5,0 Sec.
	18	5,0 Sec.	10,0 Sec.	2,5 Sec.	5,0 Sec.
	16	7,5 Sec.	15,0 Sec.	3,8 Sec.	7,5 Sec.
	8	15,0 Sec.	30,0 Sec.	7,5 Sec.	15,0 Sec.
<hr/>					
@ 44.1KHz	24	5,4 Sec.	10,9 Sec.	2,7 Sec.	5,5 Sec.
	18	5,4 Sec.	10,9 Sec.	2,7 Sec.	5,5 Sec.
	16	8,1 Sec.	16,3 Sec.	4,1 Sec.	8,2 Sec.
	8	16,3 Sec.	32,6 Sec.	8,2 Sec.	16,3 Sec.

table 1.

name*

Press DO to name your sample. This helps you to get a good overview of the samples stacked in the memory. This feature is also available when you save a sample to disk.

COUNTER

0.0s - (max. sampletime)

Sample time counter. In playback mode it displays the length of the sample and in rec. mode it displays available sample-

STATUS ready?, armed!, playing, record., looping time according to RECMODE and FREEMEM.
Read Only. Shows the current action of the sampler.

PAGE 3:

START 0.00s - (end point) Edit start point of current sample. When the sample is edited, a small part of the sample playbacks for cue listening.

FINE 0.00ms - 9.99ms Fine adjustment of start point with cue.

END 0.00s - (max. sampletime) Edit end point of current sample. When the sample is edited a small part of the sample playbacks for cue listening.

FINE 0.00ms - 9.99ms Fine adjustment of end point cue.

PAGE 4:

RECMODE mono, stereo Switch between mono- or stereo sampling. Max. sampletime will be displayed in FREEMEM according to chosen mode .

FIL-RES* 8, 16, 18, 24 bit **For disk storage only.** This is the wordwidth used when storing samples. The higher FILE resolution the higher disk storage capacity is required (See table 1).

FREEMEM* (Read Only) Displays the total available sample time in seconds according to RECMODE and installed SIMM. After power up all memory installed is cleared and available for sampling.

Max. sampletime	Mono	Stereo	Mono	Stereo
Samplerate	44.1 KHz	44.1 KHz	48 KHz	48 KHz
Standard	1,5	0,8	1,4	0,7
High Mem (51RAM) installed	3,0	1,5	2,8	1,4
Dynamic RAM installed				
1 MByte	23,8	11,9	21,8	10,9
4 MByte	95,1	47,6	87,4	43,7
16 MByte	380,4	190,2	349,5	174,8

Table 2.

REVERSE off - on If 'on' current sample is played back in reverse.

PAGE 5:

LEVEL	off - 0.0dB	Sets the playback level of selected sample.
PAN	50 L - center - 50 R	Pans the selected sample between left and right.
FADEIN	0.00s - 1.00s	Sets the fade-in time for the selected sample. This parameter should normally be set to 0.00s.
FADEOVR	0.00s - 1.00s - to end	Sets the time that the running sample continues when a retrig occurs. To avoid a doubling effect on longer samples set this parameter to 0.00s . To play the running sample through to the end, set FADEOVR to ' to end '. The fadeovr exists to avoid the annoying and unnatural sounding cut off caused by restarting a sample before it has finished playing. This feature is useful on certain drum and percussion sounds.

PAGE 6:

LOCUT*	20Hz - 1.00KHz	Low cut filter, shelving type. Provides an overall low frequency rolloff (6 dB per octave). Sets the cutoff frequency of the overall low cut filter in 1/3-octave steps. In 'loop' mode DO-button must be pressed again after editing the filters.
LO-ATT*	0.0dB - -40.0dB	The attenuation control sets the low frequency rolloff determined by LOCUT in 0.5 dB steps.
HICUT*	1.00KHz - flat	High cut filter, shelving type. Provides an overall high frequency rolloff (6 dB per octave). Sets the cutoff frequency of the overall high cut filter in 1/3-octave steps. In 'loop' mode DO-button must be pressed again after editing the filters.
HI-ATT*	-40 - 0.0 dB	The attenuation control sets the high frequency rolloff determined by HICUT in 0.5 dB steps.

PAGE 7:

TRIGGER	manual, pedal, midi*	Enables different trigger modes (playback of samples). When set to ' manual ', triggering of sample can be done by pressing the DO-button. Choose ' pedal ' to trig sample also from the pedal connector on the back panel of the M5000 (normally open contact). You can also trigger your samples from a MIDI* keyboard (see next page).
TRIGLEV	off - 0.0dB	Sets the threshold level for the audio triggering input. The fast trig will not respond to input levels below -30dB.
DEADBND	-20dB - 0.0dB	Sets the level the audio level needed to go below TRIGLEV before a new audio trig is possible. Active only when audio triggering.
RETRIG	0.03s - 1.0s - to end	A trigger mask that sets the time that must pass before a new audio trig is possible. With 'to end' selected audio retrigger is not possible before sample has ended.. Active only when audio triggering.

PAGE 8:

MIDIkey*	0 - c0 - c7 - 127	When MIDI trigger is chosen (TRIGGER=midi) the samples can be triggered from a MIDI device, i.e. a MIDI keyboard. Select a keynote on which you want the sample to respond. Note: ACTION (page 2) must be set to play!
NOTEoff*	off - on	Enables the M5000 to respond to ' note off ' MIDI command. When NOTEoff is set to off the selected sample will playback its whole length regardless of the key is released. When NOTEoff is set to on the selected sample will stop playback when the keyboard key is released.

VELsens*

off - on

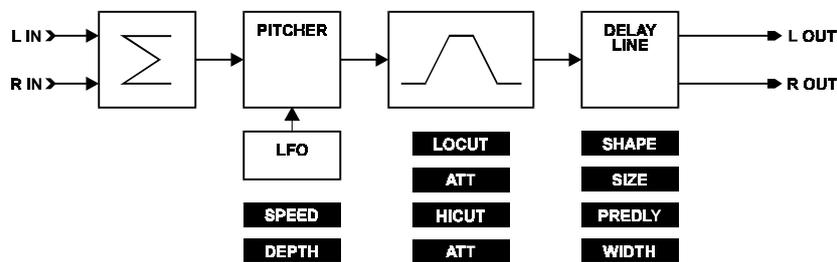
When set to '**on**' the sample level will correspond to the value transmitted from the MIDI device.

MIDI CH* omni - ch 1 - ch 16 Sets the MIDI channel on which the samples receives the MIDI commands. It must match the transmitting MIDI device. In '**omni**'-mode the samples will

* Not possible if you are running the STANDARD sampler - the parameters are adjustable but do not affect the signal. Simply install SIMM memory and all parameters will be available.

This is a brief description of the parameters of the AMBIENCE algorithm. This algorithm is based on the well-known REVERB-1 and REVERB-2 early reflection patterns with some additional parameters. The high resolution of the implemented room shapes makes it possible for simulation of small ambient rooms only by the early reflections. An obvious application could be simulation of e.g. kitchens, dining rooms, living rooms with or without furniture. In other words, more or less specialized to film applications. However, the algorithm is also capable of adding new characteristics to a recording studios ambient recordings.

The diagram below is an addition to the signal flow diagram found in the "BASIC ALGORITHMS" module, page 2.



EDIT PARAMETERS:

MIX	0 - 100 %	Sets the mix between dry and wet signal. 100 % = effect signal only. Mix can be set to 100 % globally in the G-LEVELS-menu under UTILITY. Set MIXMODE WET=MAX and all dry signals are "killed" regardless of preset mix settings.
INLEV	off - 0.0 dB.	Sets the level of the input to the algorithm in 0.5 dB steps. The function of the control is to maximize dynamic range. Please note that this control is positioned after the input PPM meter, and will not affect the input PPM reading, also if using an analog input, make the analog input adjustment in the G-LEVELS-menu under UTIL before setting this control. However, if the red overload LED flashes, turn down INLEV a couple of dBs. The control does not affect the bypassed signal level.

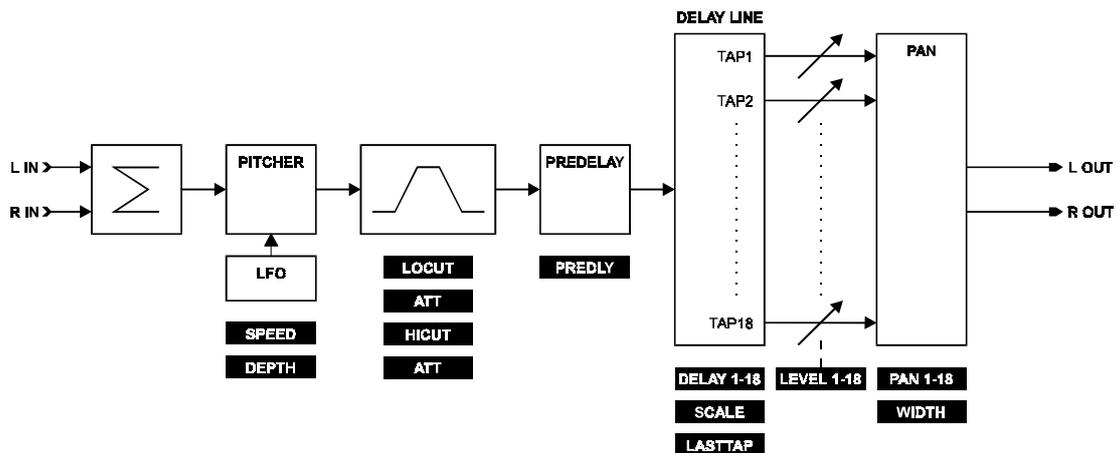
OUTLEV	off - 0.0 dB.	<p>Normally, you do not have to change the factory default setting.</p> <p>Sets the output level of the algorithm in 0.5 dB steps. The function of this control is to maximize dynamic range by allowing the ambience algorithm to output maximum signal to the DA converters. It affects the output PPM reading. There is a separate output level control for adjusting the analog output level in UTIL. Set the analog input (if you use analog input) and the INLEV adjustments before setting this control. The control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.</p>
SHAPE		<p>Room/Hall simulation/equivalent. With this control the initial pattern is chosen. Six distinctively different room shapes are available:</p>
	HALL	<p>The HALL reflection pattern is based on the acoustic properties of the Boston Symphony Hall, USA.</p>
	FAN	<p>The FAN pattern is based on the La Scala Concert Hall in Milan, Italy.</p>
	PRISM	<p>The PRISM pattern is from acoustic designers 'Golden Ratio' shoe box shaped Hall.</p>
	H.SHOE	<p>The Horseshoe shaped pattern is based on the Musikvereinssaal, Austria.</p>
	CLUB	<p>The CLUB pattern is based on the typical dimensions of a club-sized location.</p>
	SMALL	<p>The SMALL pattern is an artificially made, relatively small room. The room has been reworked to minimize some of the unfortunate coloring artifacts that would otherwise have dominated a room of this size.</p>
x SIZE	0.040 - 4.000	<p>Scales the dimensions of the simulated space depending on the SHAPE chosen. The specific room that is being simulated is scaled 1:1 at SIZE =1.00. This can then be scaled up or down (see table 1 in</p>

		REVERB-1 algorithm). Provided that the predelay setting is relatively short, the corresponding volume of the simulated space is changed radically with this control. For example; with the HALL initial pattern, the approximate room volume goes from 1.2 cubic meters to 1,280,000 cubic meters.
PREDLY	0.0 - 100.0 mS	Sets the time that passes before the first reflection appears. Maximum predelay depends on SHAPE (see table 2 in REVERB-1 algorithm text module). Increasing the predelay will change the apparent position and, to some degree, the size of the room.
WIDTH	0 - 100 %	Sets the apparent stereo width of the reflections. At '0', the reflections will appear to be coming mainly from the center (mono), whereas with WIDTH set to '100' the L/R are independent (and mono compatible).
LOCUT	20 Hz - 1.00 KHz	Low cut filter, shelving type. Provides an overall low frequency rolloff (6 dB per octave). Sets the cutoff frequency of the overall high cut filter in 1/3-octave steps.
ATT	-40 - 0.0 dB	The attenuation control sets the low frequency roll determined by LOCUT.
HICUT	1.00 KHz - flat	High cut filter, shelving type. Provides an overall high frequency rolloff (6 dB per octave). Sets the cutoff frequency of the overall high cut filter in 1/3-octave steps.
ATT	-40 - 0.0 dB	The attenuation control sets the high frequency roll determined by HICUT.
SPEED	0.100 - 10 Hz	Adding modulation to the ambience has the effect of smoothing out the frequency response, by effectively averaging out the room resonances. Note that adding even the least amount of modulation will cause the very high frequencies to diminish slightly and some detuning to occur.

DEPTH	0 - 100 %	Determines how wide a modulation (sweep) is produced.
PLDYMUL	x 1, x size	Pre-delay multiplier. When set to 'x 1' the pre-delay time will be set according to the value of the pre-delay parameter. When set to 'x size' the pre-delay will be multiplied with the SIZE parameter. In this case a scaling of the room size will automatically adjust the pre-delay accordingly.

This is a brief description of the parameters in the TAPFAC algorithm, which is short for 'Tap Factory'. With this 'factory' you are able to control up to (the very first) 18 reflection taps enabling you to produce your own unique reflection pattern. Each tap can be individually adjusted with parameters like; delay, level and pan. With 18 taps, all with different settings, you can create the most complex room simulations such as stairways, chimneys, outdoor soundfields etc.

The diagram below is an addition to the signal flow diagram found in the "BASIC ALGORITHMS" module, page 2.



EDIT PARAMETERS:

MIX	0 - 100 %	Sets the mix between dry and wet signal. 100 % = effect signal only. Mix can be set to 100 % globally in the G-LEVELS-menu under UTILITY. Set MIXMODE WET=MAX and all dry signals are "killed" regardless of preset mix settings.
INLEV	off - 0.0 dB.	Sets the level of the input to the algorithm in 0.5 dB steps. The function of the control is to maximize dynamic range. Please note that this control is positioned after the input PPM meter, and will not affect the input PPM reading, also if using an analog input, make the analog input adjustment in the G-LEVELS-menu under UTIL before setting this control. However, if the red overload LED flashes, turn down INLEV a couple of dB's. The control does not affect the bypassed signal level.

OUTLEV	off - 0.0 dB.	<p>Normally you do not have to change the factory default setting.</p> <p>Sets the output level of the algorithm in 0.5 dB steps. The function of this control is to maximize dynamic range by allowing the TAPFAC algorithm to output maximum signal to the DA converters. It affects the output PPM reading. There is a separate output level control for adjusting the analog output level in UTIL. Set the analog input (if you use analog input) and the INLEV adjustments before setting this control. The control does not affect the bypassed signal level. Normally you do not have to change the factory default setting.</p>
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PAGE 2:

SCALE	0 - 100 %	<p>Sets the relative spacing of the taps to allow the scaling of 'x SIZE' of the space created. Example: With four taps set at 11, 13, 15 and 17ms and the SCALE set at 50% the actual tap lengths are 5.5, 6.5, 7.5, and 8.5ms. This parameter is extremely useful because it changes all 18 taps simultaneously without having to do individual tap adjustments.</p>
PREDLY	0.0 - 100.0 mS	<p>Sets the time that passes before the first tap appears. Increasing the predelay will change the apparent position and, to some degree, the size of the room.</p>
WIDTH	0 - 100 %	<p>Sets the apparent stereo width of the taps. At '0' all taps will appear to be coming from the center (mono), whereas with WIDTH set to '100%' the taps appear at the L/R positions set by the PAN parameter.</p>
LASTTAP	1 - 18	<p>Selects the amount of active taps starting from 1. When set to 18 all 18 taps are active.</p>

PAGE 3:

TAP	1 -18	Selects the tap to be adjusted. Select 1 to edit the first tap. Then turn dial "A" one click to the right to edit the no. 2 etc. All 18 taps can be edited according to LASTTAP.
DELAY	0 - 624 ms	Sets the delay time for the selected TAP.
LEVEL	0 - 100 %	Sets the level of the selected TAP.
PAN	----- -----	Sets the panning of the selected TAP.

PAGE 4:

LOCUT	20 Hz - 1.00 KHz	Low cut filter, shelving type. Provides an overall low frequency rolloff (6 dB per octave). Sets the cutoff frequency of the overall high cut filter in 1/3-octave steps.
ATT	-40 - 0.0 dB	The attenuation control sets the low frequency roll determined by LOCUT.
HICUT	1.00 KHz - flat	High cut filter, shelving type. Provides an overall high frequency rolloff (6 dB per octave). Sets the cutoff frequency of the overall high cut filter in 1/3-octave steps.
ATT	-40 - 0.0 dB	The attenuation control sets the high frequency roll determined by HICUT.

PAGE 5:

SPEED	0.100 - 10 Hz	Adding modulation to the taps has the effect of smoothing out the frequency response, by effectively averaging out the resonances. Detuning of the created space will result from the use of this parameter so use judiciously.
DEPTH	0 - 100 %	Determines how wide a modulation (sweep) is produced.

This algorithm is the Digital Equalizer part of the MD2 extension TOOLBOX™.

This digital EQ features a four-band parametric EQ with high- and low-pass filters switchable to notch, shelving and cut filters. The needle sharp notch filter has a range down to 0.02 octave, the shelving filters has a variable slope ranging from gentle 3 dB/oct over 6 and 9 to 12dB/oct. Cut filters are switchable between 12dB/oct maximum flat amplitude (Butterworth) or flat group delay (Bessel) types. The parametric equalizer features a natural and well defined bandwidth behavior at all gain and width settings:

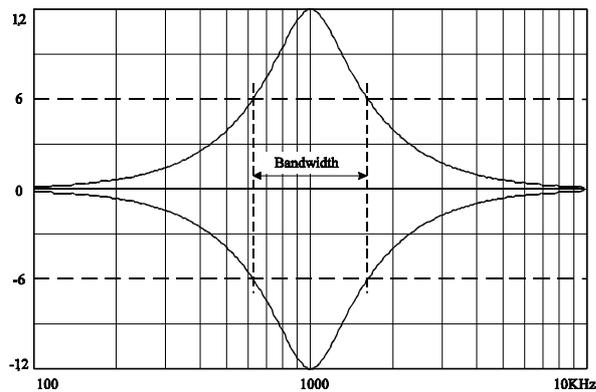


Fig.1 The bandwidth of the parametric EQ is expressed in octaves and is defined at half the EQ gain

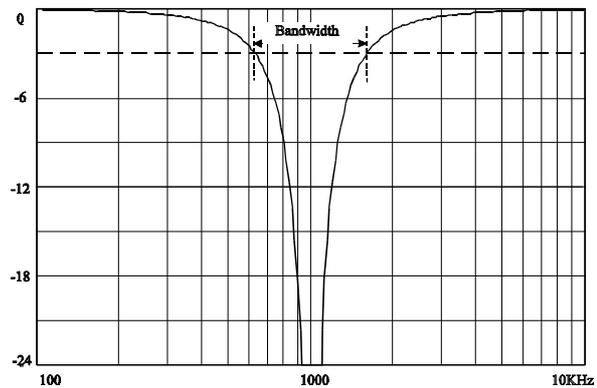


Fig.2 The bandwidth of the notch filter is defined at its -3dB points.

The shelving and parametric filters have a 100% symmetrical boost/cut response, i.e. a positive setting in one band can be canceled exactly by another with the same negative gain setting (using the same frequency and bandwidth settings - like fig. 1) .

All equalizer settings can be changed ‘on the fly’ with no unnatural audible artifacts. A fast acting morphing technique naturally transforms any EQ setting into another (including EQ type and on/off selections). The morph time is fixed.

All filters are minimum phase types - i.e. there is a unique relationship between the amplitude and the phase response of the filters. The filters are done in extended resolution implementations with active noise shaping that forces errors at the 48th bit level further towards zero.

EDIT PARAMETERS:

MIX	100 %	Locked in 100% wet mode.
INLEV	off - 0.0 dB.	Sets the level of the input to the algorithm in 0.5 dB steps. The function of the control is to maximize dynamic range. Please note that this control is positioned after the input PPM meter, and will not affect the input PPM reading, also if using an analog input, make the analog input adjustment in the G-LEVELS-menu under UTILITY before setting this control. However, if the red overload LED flashes, turn down INLEV a couple of dBs. The control does not affect the bypassed signal level. Normally you do not have to change the factory default setting.
OUTLEV	off - 0.0 dB.	Sets the output level of the algorithm in 0.5 dB steps. The function of this control is to maximize dynamic range by allowing the algorithm to output maximum signal to the DA converters. It affects the output PPM reading. There is a separate output level control for adjusting the analog output level in UTILITY. Set the analog input (if you use analog input) and the INLEV adjustments before setting this control. The control does not affect the bypassed signal level.

PAGE 2:

LO-EQ	off, on	Switches the low-EQ filter off and on.
MID-EQ1	off, on	Switches the mid-EQ1 filter off and on.
MID-EQ2	off, on	Switches the mid-EQ2 filter off and on.
HI-EQ	off, on	Switches the high-EQ filter off and on

PAGE 3:

DIAL A	DIAL B	DIAL C	DIAL D
LO-EQ	FREQ	WIDTH/SLOPE	LEVEL
par.eq	19.95Hz - 5.01KHz	0.1 oct - 4.0 oct	±12 dB
notch	19.95Hz - 5.01KHz	0.02 oct - 1.0 oct	0.0dB - off
shelve	19.95Hz - 5.01KHz	3/6/9/12 db/oct	±12 dB
cut	19.95Hz - 5.01KHz	Butterw/Bessel	

LO-EQ The low frequency filter of the 4-band equalizer. The use of this filter is determined by softdial A.

PAGE 4:

DIAL A	DIAL B	DIAL C	DIAL D
MID-EQ1	FREQ	WIDTH/SLOPE	LEVEL
par.eq	19.95Hz - 20.0KHz	0.1 oct - 4.0 oct	±12 dB
notch	19.95Hz - 20.0KHz	0.02 oct - 1.0 oct	0.0dB - off

MID-EQ1 The 1st midrange frequency filter of the 4-band equalizer. The use of this filter is determined by softdial A.

PAGE 5:

DIAL A	DIAL B	DIAL C	DIAL D
MID-EQ2	FREQ	WIDTH/SLOPE	LEVEL
par.eq	19.95Hz - 20.0KHz	0.1 oct - 4.0 oct	±12 dB
notch	19.95Hz - 20.0KHz	0.02 oct - 1.0 oct	0.0dB - off

MID-EQ2 The 2nd midrange frequency filter of the 4-band equalizer. The use of this filter is determined by softdial A.

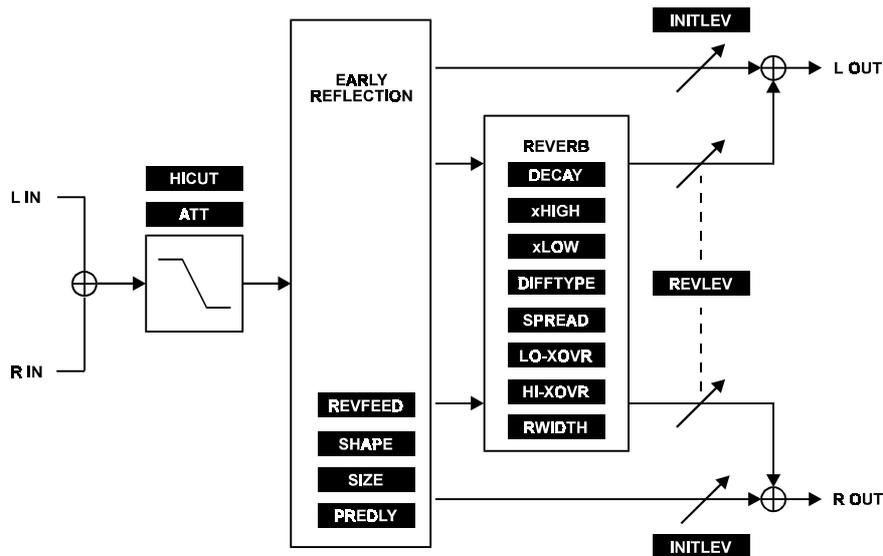
PAGE 6:

DIAL A	DIAL B	DIAL C	DIAL D
HI-EQ	FREQ	WIDTH/SLOPE	LEVEL
par.eq	501.2Hz - 20KHz	0.1 oct - 4.0 oct	±12 dB
notch	501.2Hz - 20KHz	0.02 oct - 1.0 oct	0.0dB - off
shelve	501.2Hz - 20KHz	3/6/9/12 db/oct	±12 dB
cut	501.2Hz - 20KHz	Butterw/Bessel	

HI-EQ The high frequency filter of the 4-band equalizer. The use of this filter is determined by softdial A.

The REVCORE-1 is the first in a row of new TC Reverb algorithms based on the TC CORE¹. This REVCORE-1 algo was developed specifically to perform small rooms. As a result of this, the Reverb buildup is fast, just like in smaller Rooms. Especially with percussive materials, this responsiveness is quite useful. Film and post production-work often requires use of smaller Rooms. Although the early reflection patterns bears names equivalent to the names used in Reverb 1&2 algos, the patterns are modified for use with smaller spaces.

Here is a description of the parameters dedicated to the REVCORE-1 algorithm.



EDIT PARAMETERS:

MIX	0 - 100 %	Sets the mix between dry and wet signal. 100 % = effect signal only. Mix can be set to 100 % globally in the G-LEVELS-menu under UTILITY. Set MIXMODE WET=MAX and all dry signals are "killed" regardless of preset mix settings.
INLEV	off - 0.0 dB.	Sets the level of the input to the reverb in 0.5 dB steps. The function of the control is to maximize dynamic range. Please note that this control is positioned after the input PPM meter, and will not affect the input PPM reading, also if using an analog input, make the analog input adjustment in the G-LEVELS-menu

¹ Co-efficient Optimized Room Emulation

		under UTIL before setting this control. However, if the red overload LED flashes, turn down INLEV a couple of dBs. The control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.
OUTLEV	OFF - 0.0 dB.	Sets the output level of the reverb in 0.5 dB steps. The function of this control is to maximize dynamic range by allowing the reverb algorithm to output maximum signal to the DA converters. It affects the output PPM reading. There is a separate output level control for adjusting the analog output level in UTIL. Set the analog input (if you use analog input) and the INLEV adjustments before setting OUTLEV. The control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.
DECAY	0.3 - 3.0 Sec.	Reverberation decay time.
x LOW	0.01 - 2.5 times	Relative decay time multiplier for low frequencies.
x HIGH	0.01 - 2.0 times	Multiplier for the high frequencies. If x HIGH time is set to 0.5, the HI decay time is half that of the nominal DECAY setting.
INITLEV	off - 0.0 dB.	Sets the level of the initial pattern. The purpose of this control is to balance the initial (early) reflection levels against the reverberating part of the reverb algorithm.
REVLEV	off - 0.0 dB.	Sets the level of the reflection envelope relative to the early reflections in 0.5 dB steps. If REVLEV is set to off you will hear only the initial reflections.
LM-XOVR	20 Hz - flat	Sets the crossover frequency for the x LOW decay time multiplier in 1/3-octave steps.

MH-XOVR	20 Hz - flat	Sets the crossover frequency for the x HIGH decay time multiplier in 1/3-octave steps.
SHAPE	HALL, FAN, PRISM, H.SHOE CLUB, SMALL	Room/Hall simulation/approximation. With this control the initial pattern of the reverb is chosen. In REVCORE-1, 4 distinctively different room shapes are available (See REVERB 1/2 for further information).
x SIZE	0.040 - 4.000	Scales the dimensions of the simulated space depending on the SHAPE chosen. The specific room that is being simulated is scaled 1:1 at SIZE =1.00. This can then be scaled up or down. Provided that the predelay setting is relatively short, the corresponding volume of the simulated space is changed radically with this control (See REVERB 1/2 for further information).
PREDLY	0.0 - 200.0 mS or 0.0 - 520.0 mS ²	Sets the time that passes before the first reflection appear. Maximum predelay depends on SHAPE (see table 1). Increasing the predelay will change the apparent position and, to some degree, the size of the room.

²Only if idx RAM mounted is 64K. If idx=32K max predelay will be 200.0 mS. Check your index ram in the CONFIG menu under UTILITY.

Max predelay before loosing taps (std. memory)						
@48KHz samplerate						
	Size	mS	Size	mS	Size	mS
Hall	0.50	112.4	1	40.7	1.28	0.0
Fan	0.50	146.8	1	109.5	2.47	0.0
Prism	0.50	135.4	1	86.7	1.89	0.0
H.Shoe	0.50	116.2	1	48.3	1.36	0.0
Club	0.50	133.4	1	82.7	1.81	0.0
Small	0.50	141.2	1	98.2	2.14	0.0
@44.1KHz samplerate						
	Size	mS	Size	mS	Size	mS
Hall	0.50	118.2	1	52.3	1.40	0.0
Fan	0.50	149.8	1	115.5	2.69	0.0
Prism	0.50	139.4	1	94.6	2.06	0.0
H.Shoe	0.50	121.7	1	59.3	1.48	0.0
Club	0.50	137.5	1	90.9	1.98	0.0
Small	0.50	144.7	1	105.2	2.33	0.0
Max predelay before loosing taps (w.option 'himem' memory, order# 51RAM-1)						
@48KHz samplerate						
	Size	mS	Size	mS	Size	mS
Hall	0.50	459.5	1	393.6	4.00	0.0
Fan	0.50	491.2	1	456.9	4.00	251.2
Prism	0.50	480.7	1	436.0	4.00	167.6
H.Shoe	0.50	463.0	1	400.6	4.00	26.2
Club	0.50	478.8	1	432.2	4.00	152.7
Small	0.50	486.0	1	446.6	4.00	209.9
@44.1KHz samplerate						
	Size	mS	Size	mS	Size	mS
Hall	0.50	500.2	1	428.5	4.00	0.0
Fan	0.50	534.6	1	497.3	4.00	273.4
Prism	0.50	523.2	1	474.5	4.00	182.4
H.Shoe	0.50	504.0	1	436.1	4.00	28.6
Club	0.50	521.2	1	470.5	4.00	166.2
Small	0.50	529.0	1	486.0	4.00	228.5

table 1.

REVFEED	0.0 - 100.0 mS 0.0 - 300.0 mS ³	Sets the time before the reverberating part of the signal starts to build up, relative to the early reflections PREDLY.
HICUT	500 Hz - flat	High cut filter, shelving type. Provides an overall reverb high frequency rolloff (6 dB per octave) that is well suited to make a warmer sound. Sets the cut-off frequency of the overall High cut filter in 1/3-octave steps.

³Only if idx RAM mounted is 64K. Check your index ram in the CONFIG menu under UTILITY.

ATT	-40 - 0.0 dB	The attenuation control sets the high frequency rolloff determined by HICUT.
SPREAD/DIFFTYPE	0-1	These two parameters work very close together. Here is a brief description of the two basic settings. When both parameters are set to 1, the REVCORE-1 will be very fast in its build up, and concentrated in the center. When set to 0 the tail will be a little broader. When set to either 0, 1 or 1, 0 the REVCORE-1 will displace the center a little to one of the sides.
RWIDTH	0 - 100 %	Sets the apparent stereo width of the reverberating part of the algorithm. At '0', the reverb tail will appear to be coming mainly from the center, whereas with RWIDTH set to '100' the L/R reverberators are inde-pendent.

The DYNAMIC1 algorithm is a high quality mastering Compressor/Limiter/Expander, which can be split in one, two or three stereo linked frequency bands using perfectly combining linear phase digital filters. Each band has numerous parameters for the precise tailoring of the dynamic properties of the audio signal in that particular frequency range.

SPECIAL NOTE: The flow chart shown below (fig. 1) comprises the entire Audio Signal Flow for the DYNAMIC1 algorithm from inputs to outputs. It is identical to the signal flow in all other algorithms except for the three utility parameters I/O: SOURCE, G-LEVELS: D-IN and G-LEVELS: MIXMODE have been fixed to respectively STEREO, 0.0dB and WET=MAX. Furthermore, a delay on the bypassed signal, identical to the nominal signal delay of the working DYNAMIC1 algorithm is included, so that you can make A/B comparisons or 'on the fly' bypass the dynamics processing without introducing a shift in the signal delay.

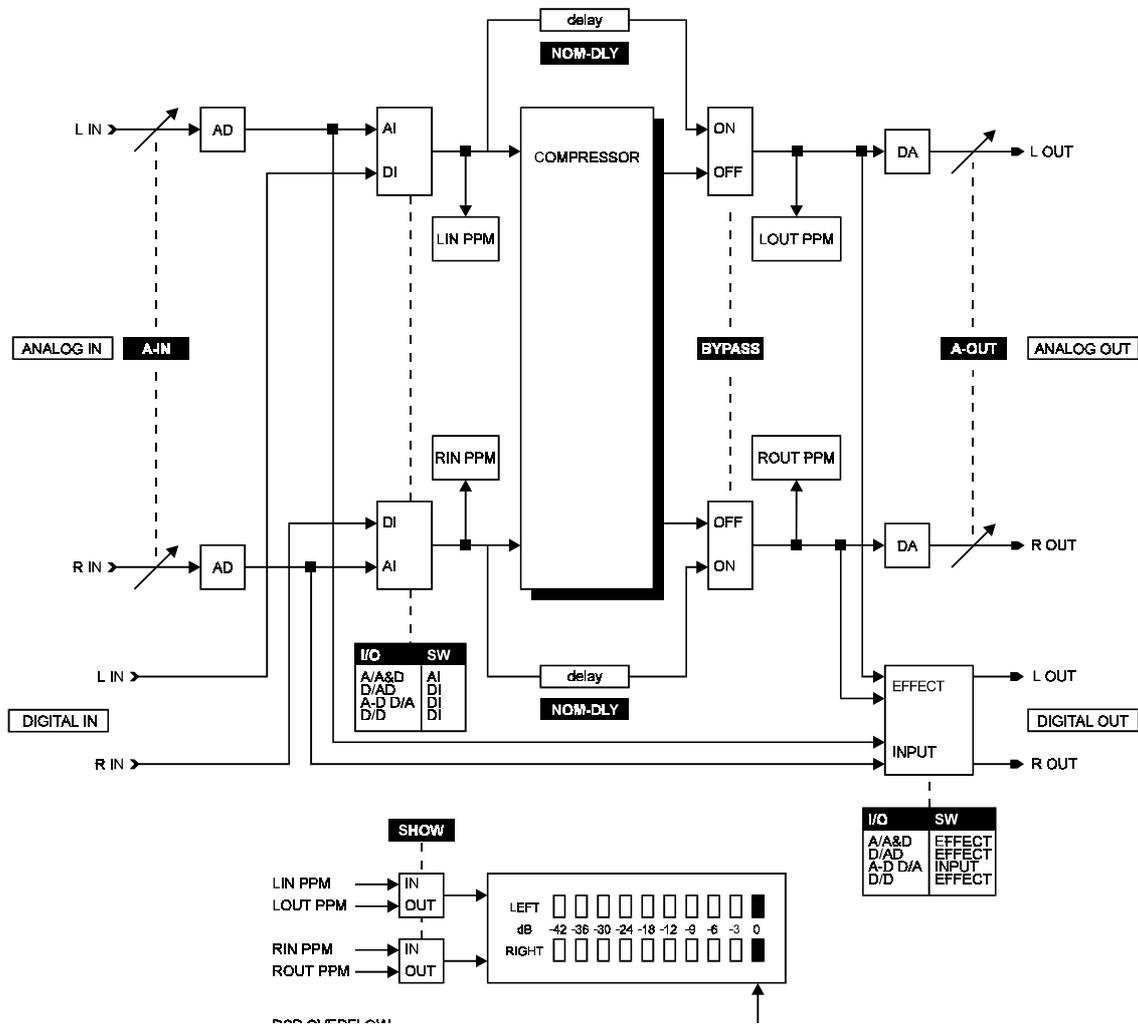


fig. 1

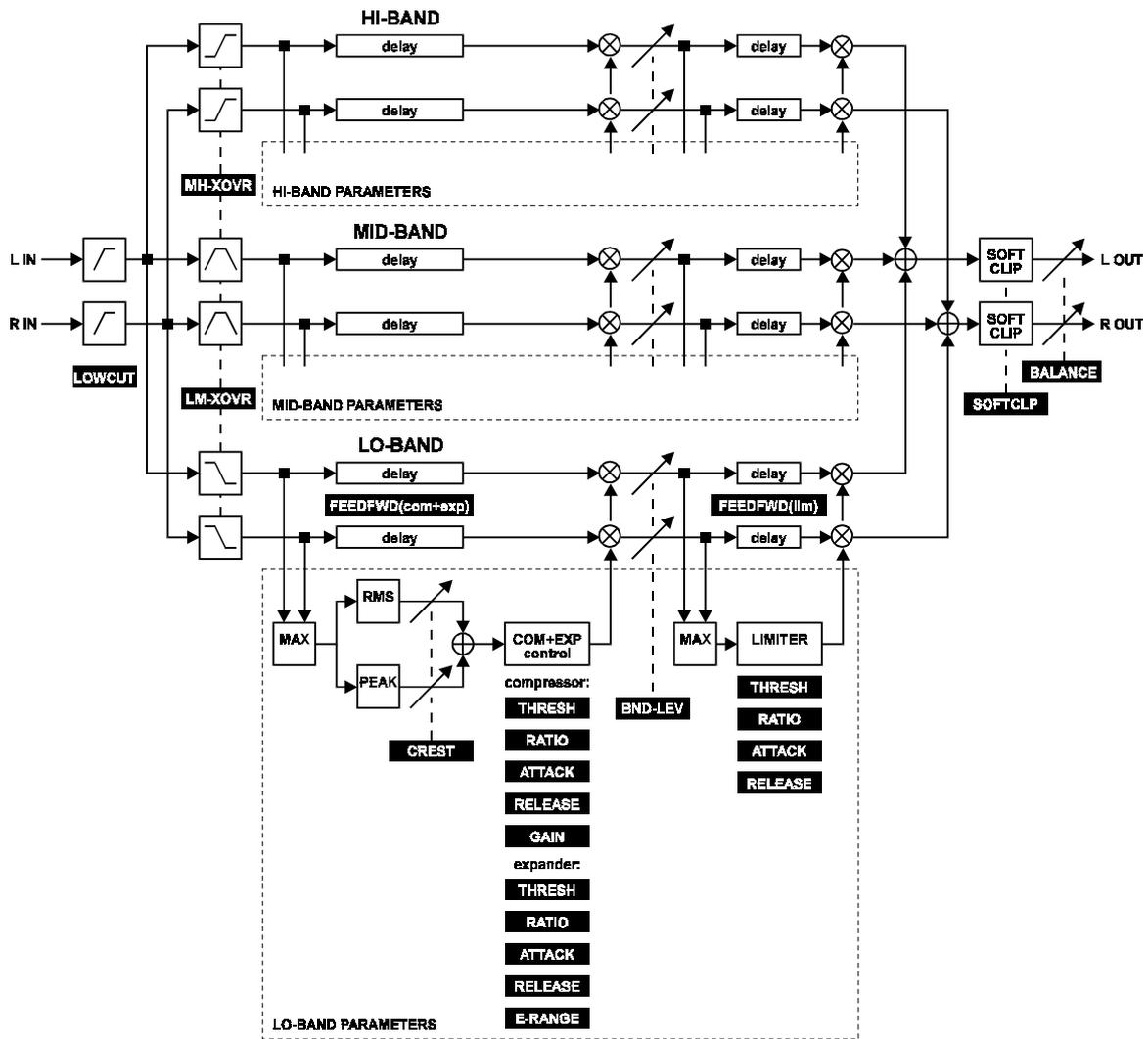


fig. 2

EDIT PARAMETERS:

As the DYNAMIC1's Compressor, Limiter and Expander can be split into 3 frequency bands it involves a lot of parameters. In order for you to have an easy user interface and quick overview of the individual gain reductions, the display will always show a gain reduction meter for each band and at the same time a selectable parameter, which can control the band individually, e.g.:

```

COMPRES L.....M.....H.....DYNAMIC1
          C-THRSH -10.0dB -12.0dB -15.0dB multibnd
  
```

Use the page buttons (11) to select either Compressor (page 4), Limiter (page 5) or Expander (page 6) and use the softdial A to select parameter.

MIX	100 %	Locked in 100% wet mode.
INLEV	off - 0.0 dB.	Sets the level of the input to the dynamic algorithm in 0.5 dB steps. The function of the control is to maximize dynamic range. Please note that this control is positioned after the input PPM meter, and will not affect the input PPM reading, also if using an analog input, make the analog input adjustment in the G-LEVELS-menu under UTIL before setting this control. However, if the red overload LED flashes, turn down INLEV a couple of dBs. The control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.
OUTLEV	off - 0.0 dB.	Sets the output level of the algorithm in 0.5 dB steps. The function of this control is to maximize dynamic range by allowing the dynamic algorithm to output maximum signal to the DA converters. It affects the output PPM reading. There is a separate output level control for adjusting the analog output level in UTIL. Set the analog input (if you use analog input) and the INLEV adjustments before setting this control. The control does not affect the bypassed signal level.
BALANCE	50 L - center - 50 R	Adjusts the balance between L and R signal.
PAGE 2:		
LOWCUT	off - 200 Hz	Filters out sub-bass frequencies and any DC component found in the audio signal. WARNING: If this is set to off you must be very sure that no DC offset is present in the signal as this can interfere with the low level function of this algorithm.
LM-XOVR	low off - 16.00 KHz	The DYNAMIC1 algorithm is separated into 3 bands. LM-XOVR sets the crossover point between low- and

MH-XOVR

mid off - 16.00 KHz

midband frequencies. When set to **low off** the algorithm is split in 2 bands.

Sets the crossover point between midrange and high band frequencies. When set to **mid off** the algorithm is a fullband compressor/limiter and is controlled with softdial D. MH-XOVR can not overlap LM-XOVR.

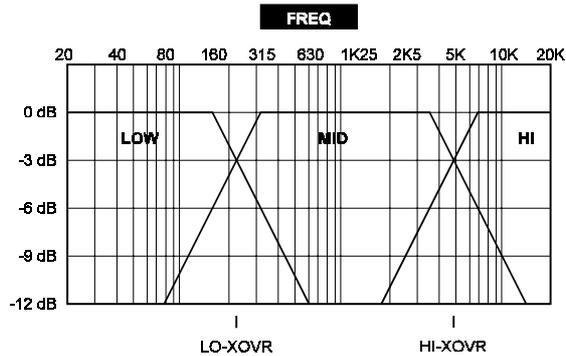


fig. 3

SOFTCLP

on - off

Soft clipping. Smoothly kills any overshoot that might occur after heavy compression or limiting. Please note that if you drive it too hard (with OUTLEV at 0dB and too much plus gain in the BND-LVL controls, you might introduce noticeable distortion, on signals with a low harmonic content and/or on very pure signals). The distortion introduced is somewhat similar to the tape saturation kind of distortion that happens in an analog tape recorder.

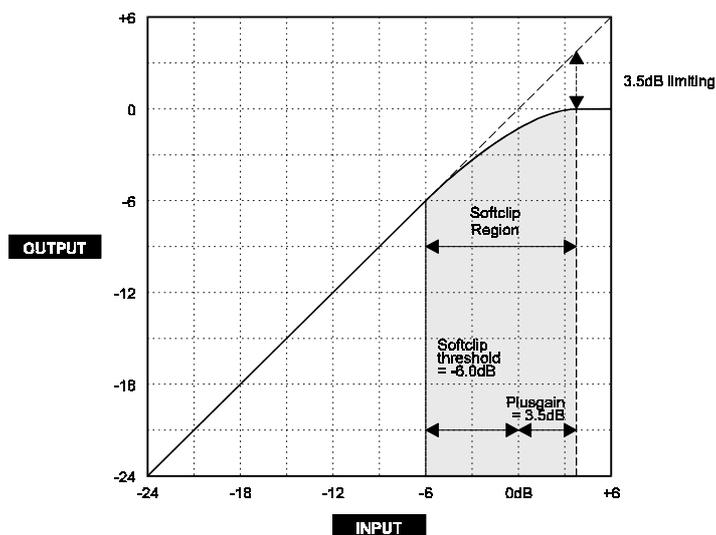


fig. 4

PAGE 3 - LEVELS:

BND-LEV	off - 0.0 dB - 12 dB	Sets the level of the individual bands.
0dB ref	-18 dB - 0.0 dB	Sets the level at which there is unity gain (output=input). In a mastering situation this value would be set between -6 dB and -10dB. For the EBU broadcast standard this would be set at -18dB.

This single control is the one to use to bring a recording into the range where the compressor is behaving in a way you want - without excessive threshold tweaking.

Note: When coming into the M5000 at the Analog inputs always set the analog input gain to make the input PPM read just below 0dB for optimum use of the A/D converter dynamic range.

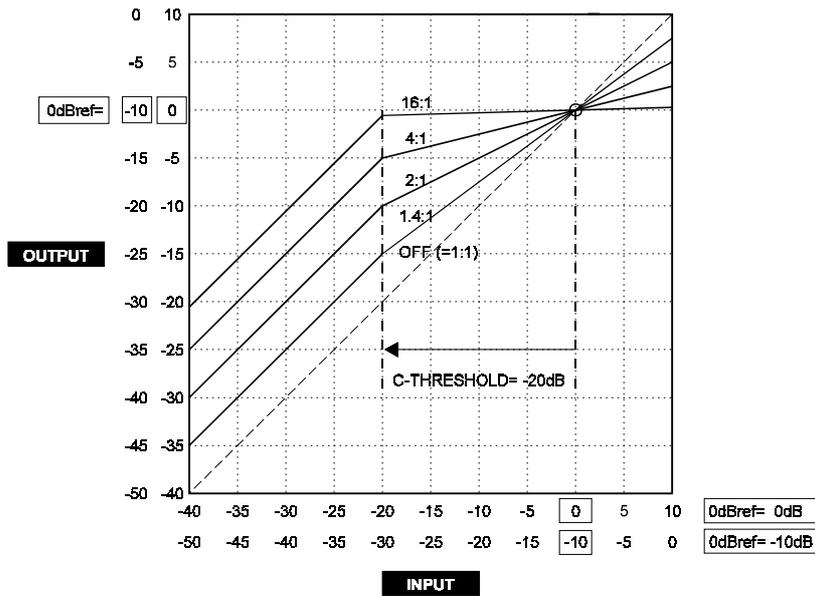


fig. 5

METERS

5 dB - 30 dB

Adjusts the full scale of the gain reduction and expansion meters. The three band meters are locked to have the same scaling. The meters will display this full scale value in 10 steps of resolution, i.e. at a setting of 5dB, each step is 0.5 dB.

Meter example:

L.....[^]f.....M.....,.....H.....[^]t...§

Each band meter is showing the gainreduction of the compressor to the right of the meter center-line and the gainreduction of the expander to the left of the centerline. Whenever the limiter of that particular band is in action a black square is shown at the end of the meter.

Illustrated above is thus an expander gainreduction of 2 dB in the low band, no action in the midrange and 2.5dB compression in the high band as well as limiting taking place.

PAGE 4 - COMPRESSOR:

C-THRSH	-40 dB - 12 dB	Compressor threshold with auto makeup function. (Very useful if there is low level at the digital inputs. You may think of this as a "drive" control for this purpose). Thresholds are relative to the '0dB ref' as shown on figure 4.
C-RATIO	off - 1.12:1 - infin:1	The ratio of the compressor can be adjusted from off (1:1) to infinite gain reduction.
C-GAIN	Read Only	Displays the auto make-up level. Lowering the threshold and/or increasing ratio on a compressor will normally result in a reduced (i.e. compressed) output level. DYNAMIC1 automatically adjusts (make up) the compressor output level in order not to lose signal level. C-GAIN will display this make up gain value, which is calculated to uphold a unity gain at a '0 dB reference' level of your choice.

Thresholds, Ratios, E-range and C-gain illustration

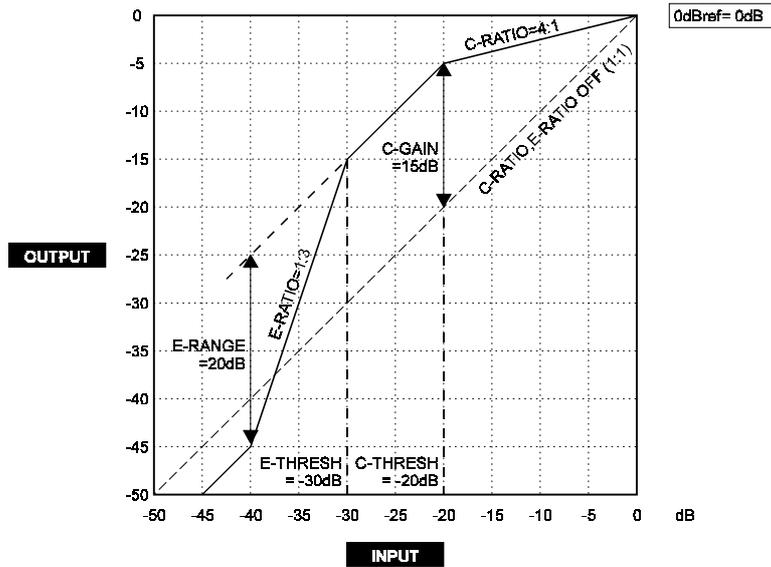


fig. 6

C-ATTCK	0.3 ms - 100 ms	Compressor attack time.
C-RLEAS	20 ms - 7.0s	Compressor release time.
FEEDFWD	0.0 ms - 25 ms	Adjusts the Compressor sidechain feed forward delay time. By slightly delaying the audio signal, the compressor has ample time in which to create the necessary level correction. To take full advantage of the digital properties of DYNAMIC1 this value should be equal to or greater than the C-ATTCK value. If faster processing is a priority this value may be set to 0.0 ms and the compressor will behave as a standard analog compressor.
		NOTE: The overall NOM-DLY parameter (page 7) must <u>always</u> be set equal to or greater than the FEEDFWD parameter found in the COMPRESSOR or LIMITER which ever is higher.
CREST	Peak - (x) dB - RMS	Adjusts the Crest Factor which determines whether the compressor shall react on peak-levels, RMS-levels or something in between, according to the adjusted C-THRSH, e.g. with a setting of 12dB, the compressor will respond to the

RMS of the input plus peaks that are 12dB higher than the current RMS value. The Root Mean Square has been found to correspond very well to our perception of level with total mixes and smoothly changing single sources. However, with more percussive types of materials you would go for a more peak oriented control of the compressor with a lower dB setting or PEAK only.

PAGE 5 - LIMITER:

L-THRSH -12 dB - 0.0 dB

Limiter threshold. The limiter is meant to be a brickwall type to prevent unintentional compressor overshoots from causing full-scale overloads. Its threshold is thus referring to digital full-scale as is the overall softclipper function.

The '0dB ref', the BND-LEV and the OUT-LEV parameters all affect 'how hard' you hit the limiter. A normal setting for CD master processing would be a few dBs down, whereas an EBU broadcaster would set L-THRESH as low as -12dB (according to the R68 recommendations.)

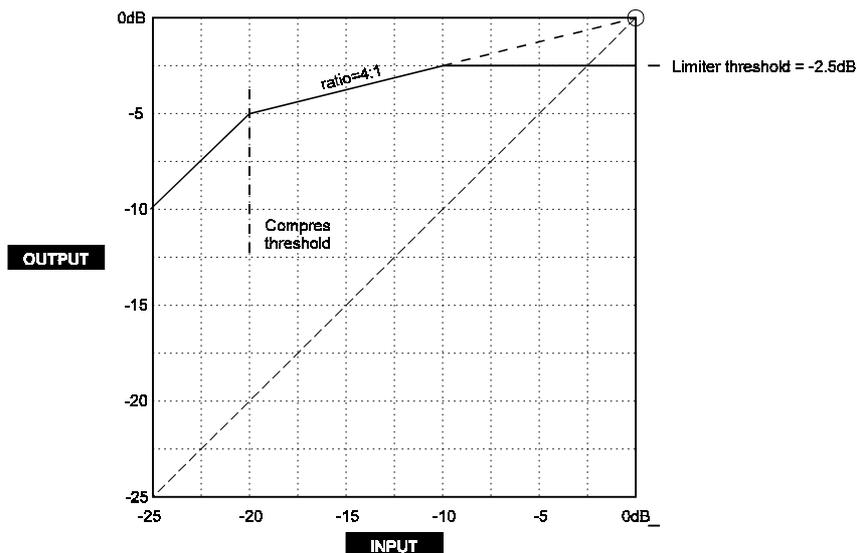


fig. 7

L-RATIO	off - infin:1	Gain reduction ratio.
L-ATTCK	30 s - 10 ms	Limiter attack time.
L-RLEAS	20 ms - 7.0 s	Limiter release time.
FEEDFWD	0.0 ms - 25 ms	Adjusts the Limiter sidechain delay time. By slightly delaying the audio signal, the limiter has ample time in which to create the necessary level correction. To take full advantage of the digital properties of DYNAMIC1 this value should be equal to or greater than the L-ATTCK value. If faster processing is a priority this value may be set to 0.0 ms and the limiter will behave as a standard analog limiter. In this case overshoot is not always suppressed

NOTE: The NOM-DLY parameter (page 7) must always be set equal to or greater than the FEEDFWD parameter found in the LIMITER or COMPRESSOR which ever is higher.

PAGE 6 - EXPANDER:

E-THRSH	-94 dB - 1.5 dB	Expander threshold.
E-RATIO	off - 1:infin	Expander ratio.
E-ATTCK	0.3 - 100 ms	Expander attack time.
E-RLEAS	20 ms - 7.0 s	Expander release time.
E-RANGE	-40.0 dB - 0.0 dB	Expander range.

PAGE 7:

PAR-LNK	off - on	Links the parameters found on any given PAGE (except '0dB ref' and METER on page 3, which always are linked) to each other in order to have common control of the bands. With LINK on any of the parameters can be adjusted and the two other bands will follow.
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NOM-DLY

0.0 ms - 25 ms

Adjusts the nominal delay common to all bands. This acts as a DDL for the full audio spectrum.

WARNING: The NOM-DLY should not be set lower than either of the FEEDFWD parameters found in the compressor or limiter pages as this parameter allows the FEEDFWD parameters to function as intended. To do so will disable the function of those parameters.

Working with the DYNAMIC1 algorithm has proven to be a very powerful tool when it comes to CD mastering and staying in the digital domain. Not only have the CD mastering plants had a very useful tool to their daily work, but also the recording studios have had great use of the DYNAMIC1 algorithm in order to deliver an even more optimized master to CD mastering plants - leaving their job a lot easier and quicker. However, with the use of the DYNAMIC1 algorithm some wishes for specific functions arose, such as equalization. Normally, one had to do this by connecting an external device - often having to convert back into the analog domain.

Digital Equalizer with parametric, notch, soft shelving and cut EQ-types.

Quantization and selectable Dithering types to 8, 12, 16, 18, 20, 22 and 24 bit levels.

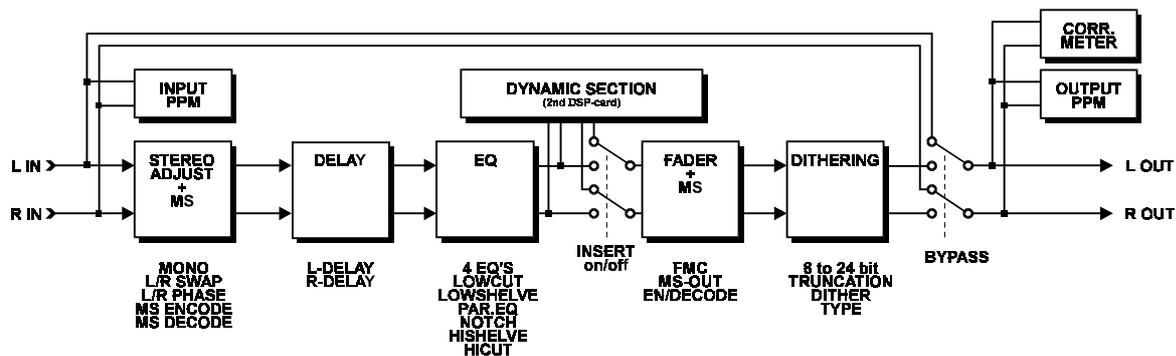
High resolution level and correlation meters.

Stereo adjust facilities with variable mono, balance, channel & phase swaps.

Digital Fading with contoured frequency corrections at lower levels.

These functions and others are implemented in this TOOLBOX™ algorithm, which is a separate algorithm that will run standalone on a DSP engine or run concurrently with e.g. the DYNAMIC1 algorithm when two DSP engines are available.

One major difference from the other M5000 algorithms is that it provides an **internal** digital insert point to which any other DSP-module can be routed. This makes it possible to run e.g. the DYNAMIC1 algorithm in conjunction with the TOOLBOX™ in a complete dynamics mastering system.



DIGITAL EQUALIZER:

The digital EQ features a four-band parametric EQ with high- and low-pass filters switchable to notch, shelving and cut filters. The needle sharp notch filter has a range down to 0.02 octave, the shelving filters has a variable slope ranging from gentle 3 dB/oct over 6 and 9 to

12dB/oct. Cut filters are switchable between 12dB/oct maximally flat amplitude (Butterworth) or flat group delay (Bessel) types. The parametric equalizer features a natural and well defined bandwidth behavior at all gain and width settings.

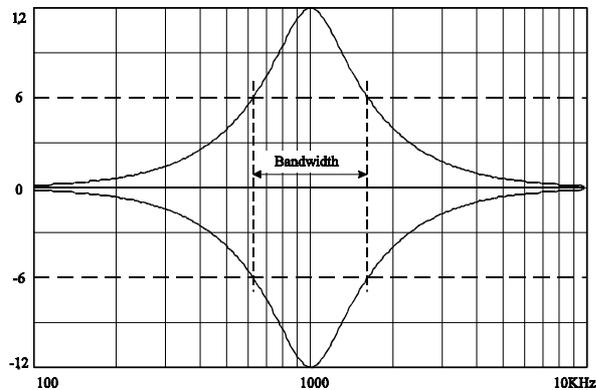


Fig.2 The bandwidth of the parametric EQ is expressed in octaves and is defined at half the eq gain

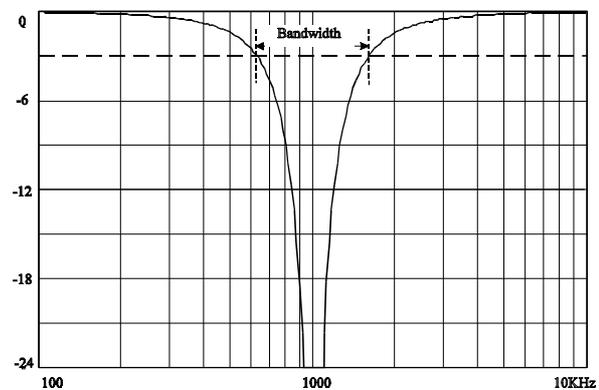


Fig.3 The bandwidth of the notch filter is defined at its -3dB points.

Shelving and parametric filters are with a 100% symmetrical boost/cut response, i.e. a positive setting in one band can be canceled exactly by another with the same negative gain setting (using the same frequency and bandwidth settings).

All equalizer settings can be changed ‘on the fly’ with no unnatural audible artifacts. A fast acting morphing technique naturally transforms any eq setting into another (including EQ type and on/off selections). The morph time is fixed.

All filters are minimum phase types, i.e. there is a unique relationship between the amplitude and the phase response of the filters.

The filters are done in extended resolution implementations with active noise shaping that forces errors at the 48th bit level further towards zero.

DITHERING:

As all processing inside the M5000 is done with a higher bit resolution than e.g. a CD or a DAT normally is capable of storing, when leaving the M5000, we are normally faced with the fact that we have too many bits. Just throwing away the bits e.g. below the 16th bit level, which will cause a graininess in the audio (and a quite objectionable distortion at low signal levels). If instead, a more intelligent form of ‘throw away bits’ processing is used, it is possible to eliminate these artifacts, and to some extent, it is even possible to obtain an audio resolution exceeding the 16 bits of the target storage medium.

The technique is simply to add a very slight amount of well controlled noise to the audio signal. This added noise will then cause the otherwise very signal dependent error signal (the thrown away bits) to lose all relations to the audio signal itself, i.e. the distortion is turned into signal independent noise. If we look at the resulting behavior of the least significant bits we may realize that they suddenly become very busy. In fact, on the average, they will tend to

represent the original 24 bit signal exactly. That is, suddenly, it is possible to pass signals below the 16th. bit level. Or put popularly, we are trading a highly unmusical graininess and level distortion for a much less noticeable noise and get an improved reconstruction of the original signal. This process is popularly called dithering. Further, a shaping of the added noise is possible. The TOOLBOX™ features 2 types of dithering: The TDF Triangular Probability Density Function, which is a flat power spectrum dithering type, and a High Frequency shaped TDF noise, that has a 5-6 dB less apparent added noise. Which one is the best depends on the program material, however in general, the HP-TDF is recommended.

METERS:

As the meters on the M5000 front panel obviously are too rough for monitoring the signal levels, a special high resolution level meter has been implemented. The meter has several features such as switchable range and ticks (dB marks) for easy monitoring of the critical levels. Maximum peak hold will display the highest peak that occurred or auto release will display the peak momentarily.

STEREO ADJUST:

Different stereo adjust parameters enables you to fine adjust the balance between left and right. You can even swap left and right channel which can be a difficult operation in the digital domain. MS encoding/decoding and mono addition are other parameters that might be useful within the digital mastering domain. Phase problems can be fixed by the special phase control.

FADING:

Digital fading is something you normally would do in your editing system, however for optimizing recordings between e.g. 2 DAT players fading possibilities are rarely available. With the TOOLBOX™ fading in this situation is readily possible.

There is even a unique fading feature - that is not readily available otherwise in a mastering plant - namely a selectable **Fletcher-Munson** corrected fade pattern - named FMC as parameter. As shown below on fig. 4, Fletcher and Munson, established a set of equal loudness contours for the human hearing, i.e. our normal experience of the loudness as a function of frequency at various levels. The term Phon is used to express our experience of loudness relative to 1 KHz.

The Fletcher-Munson Correction parameter adds a frequency contouring that is linked to the fade dB value. The correction chosen in the TOOLBOX™ comes in action at fader values below -20dB only and makes the loss, as you fade out, of both low and high frequencies much less apparent. Maximum correction added happens at -60dB and is close to +20dB relative to 3KHz. By fading with this pattern the program material seems to be more linear and pleasant to listen to during a fade out, instead of the usual 'thinning out' as you fade out.

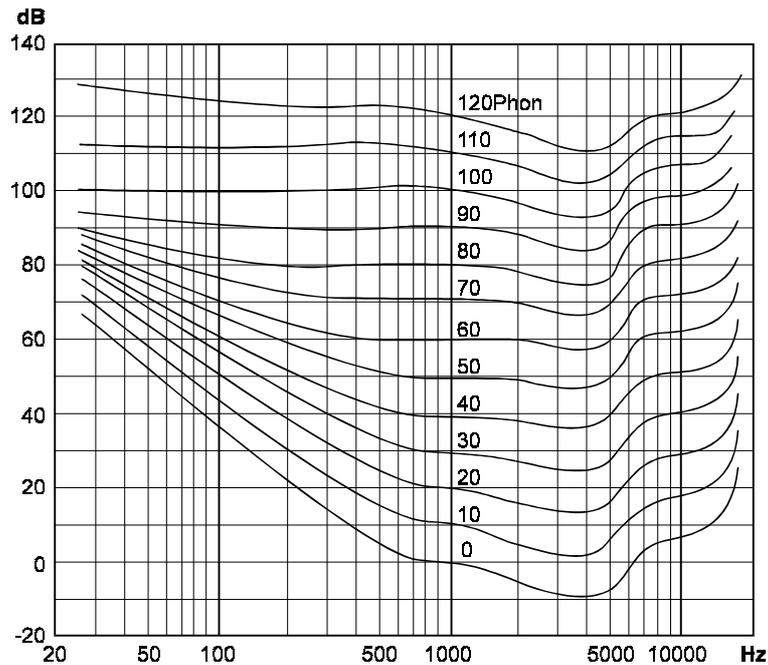


Fig. 4 The Fletcher-Munson Equal Loudness Contour Curves

EDIT PARAMETERS:

MIX	100 %	Locked in 100% wet mode.
INLEV	off - 0.0 dB.	Sets the level of the input to the TOOLBOX™ algorithm in 0.5 dB steps. The function of the control is to maximize dynamic range. Please note that this control is positioned after the input PPM meter, and will not affect the input PPM reading, also if using an analog input, make the analog input adjustment in the G-LEVELS-menu under UTILITY before setting this control. However, if the red overload LED flashes, turn down INLEV a couple of dB's. The control does not affect the bypassed signal level. Normally you do not have to change the factory default setting.
OUTLEV	off - 0.0 dB.	Sets the output level of the algorithm in 0.5 dB steps. The function of this control is to maximize dynamic range by allowing the TOOLBOX™ algorithm to output maximum signal to the DA converters. It affects the output PPM

reading. There is a separate output level control for adjusting the analog output level in UTILITY. Set the analog input (if you use analog input) and the INLEV adjustments before setting this control. The control does not affect the bypassed signal level.

LOWCUT	on/off	Filters out sub-bass frequencies and any DC component found in the audio signal. The filter crossover frequency (-3dB) is fixed at 2.5Hz and is using a non-obtrusive 6 dB/oct slope type filter. If needed, more steep LOWCUT functions can be found in the eq section.
--------	--------	--

PAGE 2:

METERS	(display) input, output	Selects whether the high resolution level meter is showing input- or output level.
RANGE	18 dB, 36 dB, 72 dB	Determines the resolution of the level meter. The full-scale range is selectable from 0 dB to either -18, -36 or -72 dB (see fig. 5-7).
TICKS	none, 6, 9 or 12 dB	A gradation to the level meter gives a quick overview of the level status.
HOLD	none, max, auto	Indicates Peak hold. If set to max or auto the peak hold either freeze the reached maximum level or if set to auto it will hold the peak momentarily.

PAGE 3, LEVEL METER

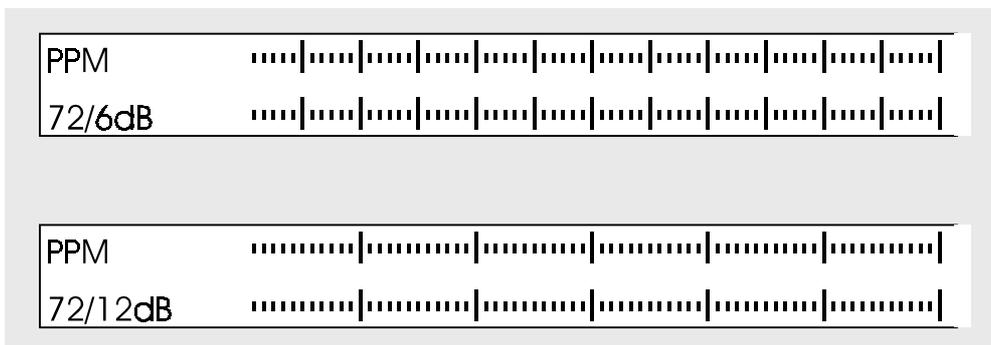


Fig. 5 The Meter range at 72 dB with the different ticks.

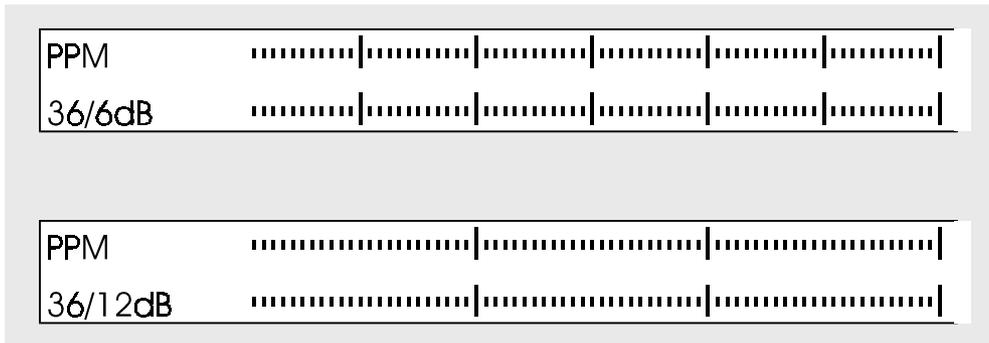


Fig. 6 The Meter range at 36 dB with the different ticks.

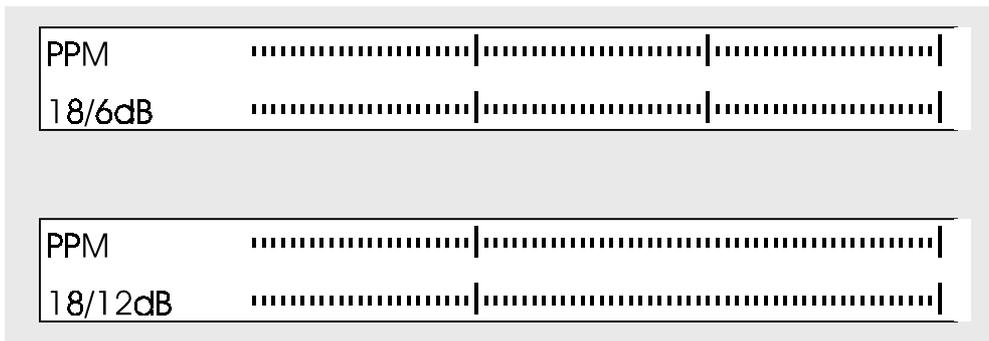
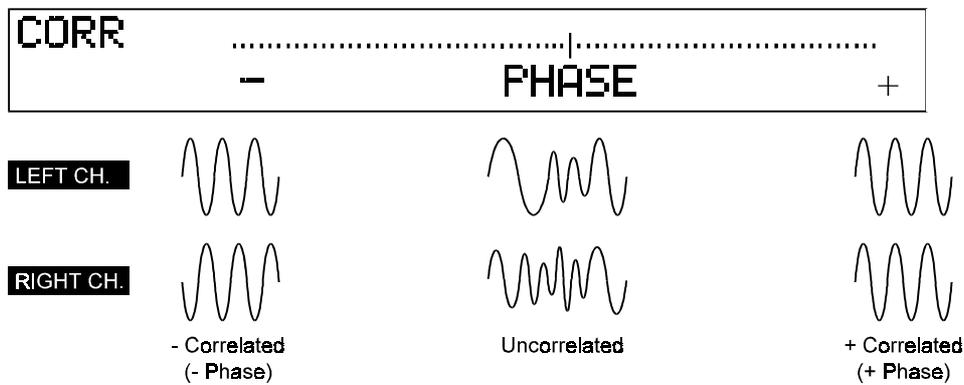


Fig. 7 The Meter range at 18 dB with the different ticks.

PAGE 4, CORRELATION METER:



The correlation meter is displaying the phase coherence within the stereo signal. A true stereo signal (uncorrelated) should act around 0 moving towards +correlation. The more +correlation the more identical (mono) are the left and right channels.

PAGE 5:

FADER -80 dB to 0 dB The high resolution fader. Turn softdial A to manually fade in or out.

FMC off, on Use the Fletcher-Munson corrected fade pattern by setting FMC to 'on'.

PAGE 6:

MS-INPUT -180deg - off - 180deg MS rotation of the input signal. Can be used for L/R conversion of a MS coded signal or for coding a L/R signal to M/S signal, both when set at +45. Adjusting the angle from +45 downward toward 'off' reduces the S-level, whereas increasing the angle in the range +45 toward +90 decreases the M-level.

MS-OUTPUT -180deg - off - 180deg MS rotation of the output signal. (After insert). Can be used for L/R conversion of a MS coded signal or for coding a L/R signal to M/S signal, both when set at +45.

PAGE 7:

BALANCE -3.0 dB to +3.0 dB Fine adjustment of the balance between left and right channel in 0.1 dB steps.

MONO 0 % - 100 % Increases the center focusing of the signal by adding L to R, R to L leakage.

LR-SWAP off, on Swaps the left and right channel.

PHASE L+R+, L-R+, L+R-, L-R- Adjust the phase between left and right channel.

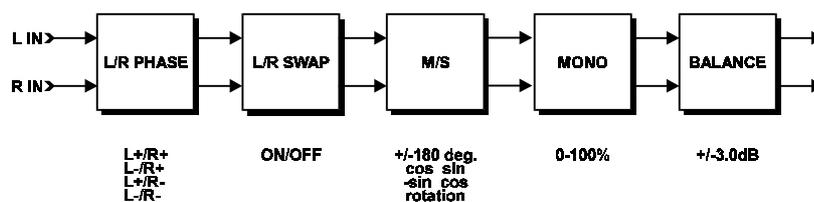


Fig. Signal flow of the input stereo adjustment parameters.

PAGE 8:

L-DELAY 0.0 ms - 300.0ms Individual delay for left channel. Can be adjusted in 0.1 ms steps.

R-DELAY	0.0 ms - 300.0ms	Individual delay for right channel. Can be adjusted in 0.1 ms steps.
INSERT ⁴	off, on	This enables you to internally insert a 2nd DSP engine in the M5000 main-frame i.e. running the DYNAMIC1 algorithm. The insert point of the TOOLBOX will always send on the digital audio bus. The INSERT parameter determines whether the TOOLBOX algorithm shall the returned signal (on) of not (off). When set to 'on' the inserted DSP's I/O configuration must be set to INSERT in the UTILITY menu.

PAGE 9:

LO-EQ	off, on	Switches the low eq filter off and on.
MID-EQ1	off, on	Switches the mid-eq1 filter off and on.
MID-EQ2	off, on	Switches the mid-eq2 filter off and on.
HI-EQ	off, on	Switches the high eq filter off and on

PAGE 10:

DIAL A	DIAL B	DIAL C	DIAL D
LO-EQ	FREQ	WIDTH/SLOPE	LEVEL
par.eq	19.95Hz - 5.01KHz	0.1 oct - 4.0 oct	±12 dB
notch	19.95Hz - 5.01KHz	0.02 oct - 1.0 oct	0.0dB - off
shelve	19.95Hz - 5.01KHz	3/6/9/12 db/oct	±12 dB
cut	19.95Hz - 5.01KHz	Butterw/Bessel	

LO-EQ The low frequency filter of the 4-band equalizer. The filter type is determined by softdial A.

PAGE 11:

⁴ Older DSP engines needs the MULTIBUS upgrade. In case of problems please contact your dealer or TC Headoffice in Denmark.

DIAL A	DIAL B	DIAL C	DIAL D
MID-EQ1	FREQ	WIDTH/SLOPE	LEVEL
par.eq	19.95Hz - 20.0KHz	0.1 oct - 4.0 oct	±12 dB
notch	19.95Hz - 20.0KHz	0.02 oct - 1.0 oct	0.0dB - off

MID-EQ1 The 1st midrange frequency filter of the 4-band equalizer. The filter type is determined by softdial A.

PAGE 12:

DIAL A	DIAL B	DIAL C	DIAL D
MID-EQ2	FREQ	WIDTH/SLOPE	LEVEL
par.eq	19.95Hz - 20.0KHz	0.1 oct - 4.0 oct	±12 dB
notch	19.95Hz - 20.0KHz	0.02 oct - 1.0 oct	0.0dB - off

MID-EQ2 The 2nd midrange frequency filter of the 4-band equalizer. The filter type is determined by softdial A.

PAGE 13:

DIAL A	DIAL B	DIAL C	DIAL D
HI-EQ	FREQ	WIDTH/SLOPE	LEVEL
par.eq	501.2Hz - 20KHz	0.1 oct - 4.0 oct	±12 dB
notch	501.2Hz - 20KHz	0.02 oct - 1.0 oct	0.0dB - off
shelve	501.2Hz - 20KHz	3/6/9/12 db/oct	±12 dB
cut	501.2Hz - 20KHz	Butterw/Bessel	

HI-EQ The high frequency filter of the 4-band equalizer. The filter type is determined by softdial A.

PAGE 14:

QUANTIZ	8,12,16,18,20,22,24 bit	Bit quantization of output. Select the proper bit resolution that suits your purpose. Dither level is automatically adjusted to match the chosen output resolution, but, to actually dither the output, DITHER TYPE should be set different from 'none'. Please note that the dither and quantization levels affects both the Analog and the Digital outputs.
DITHER TYPE	none, TDF, HP-TDF	Dithering type selects between none (pure truncation, w. no dither added) TDF dither and high frequency contoured HP-TDF dither.

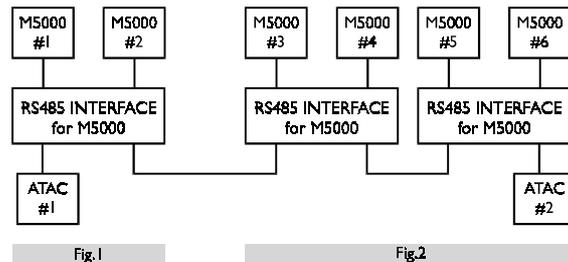
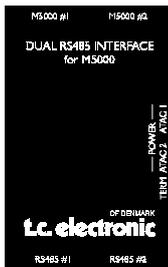
CONFIGURATION

This section contains text modules concerning configuring either software or hardware applications. When software is released or a new module card is purchased, refer to this section in order to install it properly. The section contains the following text modules:

- ✓ M5000 APPLICATIONS
- ✓ SOFTWARE INSTALLATION
- ✓ HARDWARE INSTALLATION
- ✓ OPTION INSTALLATION
- ✓ HIGH MEMORY OPTION
- ✓ SIMM PACK INSTALLATION

MULTIPLE ATAC/M5000/M5000X SETUP

The ATAC-remote system is capable of controlling multiple M5000/X main frames. The link that enables the data communication is the **TC DUAL RS485 INTERFACE for M5000/X**, also referred to as the **MULTAC**, and the proprietary **TC Network Protocol**. The RS485 INTERFACE and a connection example is illustrated below:



With one RS485 INTERFACE (MULTAC) you are able to connect one or two M5000/X frames. The M5000/X's are to be connected to the M5000/X #1 and #2 plugs (Fig.1).

The RS485 plugs each connect to either one ATAC or is used to loop on to the next MULTAC box (Fig.2). Additionally, you are able to connect two ATACs using both RS485 sockets.

When connecting two ATACs, in a multiple M5000/X/ATAC setup, you gain the advantage that two users can communicate simultaneously. Please note that two users cannot access the same mainframe at once.

The MULTAC also has DC plug connections for the ATAC power supplies.

Cables:

You may substitute the cable, connecting the ATAC to the MULTAC, with a "MIDI Plus" cables (5 pins with shield).

A 7 pin cable with shield must be used when connecting the M5000/X with the MULTAC.

Note: To ensure a safe data transferal, please follow the cable length requirements stated below:

1. When only ONE M5000X is connected use max. 10m/33ft of remote cable.
2. In a MULTIPLE M5000/X/ATAC setup, lengths of the cables connecting the ATAC to the MULTAC can be extended to 100m/328ft. Maximum lengths may not exceed 100 meters/328 ft.

Here is the procedure for updating the application software in the M5000. You need to get to the special M5000 Setup Utility menu to accomplish this.

1. Turn the power off the M5000.
2. Press the **BYPASS** button (**22**) while switching power on again. Hold it for a few seconds. The **M5000 Setup Utility Menu** will appear.
3. With the **PROGRAM DIAL** (**17**) you choose the appropriate option:

LOAD DISK/CARD When **DO** is pressed the directory of the inserted floppy disk or memory card will appear. Scroll through the files with **PROGRAM DIAL**. Select the file by pressing **DO** and confirm the choice by pressing **DO** again. Software from floppy disk or memory card will be loaded. The **UNDO** button works as a cancel button and returns you to the Setup Menu.

SAVE DISK/CARD Saves the application software to a formatted floppy disk or memory card. By pressing **DO** you enter the name menu which enables you to give the application software a file name on the disk/card. On the top line a file name is displayed with a cursor under the first character - ready for editing.

SOFT DIAL A:	CURSOR	Moves the cursor forward or backwards through the name. The name can have a maximum of 8 characters.
SOFT DIAL B:	LETTERS	Selects a letter from A to z and inserts it in the name over the cursor.
SOFT DIAL C:	FIGURES	As letters but numerical from 0 to 9.
SOFT DIAL D:	SYMBOLS	Inserts symbols instead of characters, e.g. blank or space is a symbol found here.

(Press **DO** to confirm file name.)

LOAD MIDI Enables you to receive software updates from another M5000 (See **SAVE MIDI**) or you can use TC's **M5DUMP** software package which enables you to dump software to the M5000 via an IBM™ compatible PC with a MIDI interface installed. This package is available from TC's bulletin board free of charge for M5000 **USER CLUB** members and instructions are implemented with the program.

- SAVE MIDI** Enables you to dump the software from one M5000 to another. Connect this (master) M5000's MIDI out to another (slave) M5000's MIDI in. Select 'SAVE MIDI' in order to transmit. The Slave M5000 must be set at 'LOAD MIDI'. Press DO at the master first and then press DO at the slave. An ERROR detector will inform you if any errors arises during MIDI dump. If any errors are detected it is recommended that you repeat the procedure. It is of course also possible to save the application software over MIDI to an IBM™ compatible PC if you haven't a floppy disk instal-led.
- SERIAL #** Read Only parameter. Shows the topical serial number of the M5000 and the BIOS version no. If the FLASH EPROM is a 2 Megabit size it is also shown here. If not - it is a 1 Megabit.
- FORMAT CARD** Formats a never used memory card or erases the existing files. Insert the unformatted card or a card you want to erase that supports the JEIDA PCMCIA standard. A 64 Kb memory card will be able to hold over 2000 programs. Press DO and the display will tell you that all data will be erased on the card. Confirm by pressing DO or abort by pressing UNDO.
- FORMAT DISK** Formats a floppy disk with the IBM™ compatible format. After format there will be 1.44 MB available on disk. This means that if the application software is stored on disk there are room for over 50,000 programs. Press DO and the display will tell you that all data will be erased on the floppy disk. Confirm by pressing DO or abort by pressing UNDO.

When the time comes when you want to upgrade your M5000 with additional modules in order to run more than one effect simultaneously, the procedure is as follows:

1. Switch the machine OFF and disconnect the main power cord.
2. Remove the DUM-1 option plate(s) or module by loosening the 2 screws.

As the M5000 modules are sensitive to static electricity, certain precautions must be taken to protect them from damage during storage and handling.

STORAGE

Non-mounted modules should always be stored in anti-static shielding bags.

GENERAL HANDLING

When inserting or removing any modules, avoid touching the circuit board by handling only the rear panel of the module. Modules should always be placed in either an M5000 or in a shielding bag. To minimize the static potentials that can cause damage to the electronic circuits, you should observe precautionary grounding techniques such as touching a grounded M5000 Audio Frame immediately before inserting modules.

REMOVING MODULES

Before removing any module from your M5000, switch off the power and unplug the main power cable. Unplug all other connections from the module before unscrewing the two screws securing the module in the Mainframe. When removing a module from an M5000, the module should be mounted directly in another M5000 or placed in an anti-static shielding bag.

MOUNTING MODULES

Before mounting modules in your M5000, switch off the power and unplug the main power cable. Remove the dummy-panel or original module from the slot where you want to install the module. The module should then be removed from the shielding bag and mounted directly in the M5000 Audio Mainframe by handling the rear panel of the module only. Avoid touching any components on the PCB-Board.

3. Set the DIP-switches on the module cards as shown below.

	Dip 1	Dip 2	Dip 3	Dip 4	Addr.
1. DSP	off	off	off	off	0
2. DSP	off	on	off	off	2
3. DSP	off	off	on	off	4
4. DSP	off	on	on	off	6

	Dip 1	Dip 2	Dip 3	Dip 4	Addr.
1. ADDA	on	off	off	off	1
2. ADDA	on	on	off	off	3

4. Insert the module as shown in figure 1 below.



fig. 1

AD/DA cards are always mounted as far away as possible from the power supply!

The module cards will fit in the module guides inside the M5000 frame. **It is important that the modules are mounted correctly in these blue guides** to ensure proper connection to the buss. It is recommended that you use a powerful light source in order to see properly inside the frame. Improper connection may cause serious damage to the modules.

5. Fasten the module with the two screws and connect the cables.
6. The module cards will be initialized during the next power up.
7. If there are problems e.g. the cards are not recognized by the M5000 frame, please check once again - especially the address settings.

In Appendix C you can find a self-test procedure to see if the M5000 has found the cards at the proper addresses and if the M5000 is working alright.

You have the facility to try the newest options within a certain time limit (normally 100 hours). First of all you need to install the new application software which is described in the 'SOFTWARE INSTALLATION' module in this section.

The option you want to install is in fact already in the M5000. However, a 20 character license code and an 8 character subcode are needed to access the option. This is done by writing a code in the M5000 generated only by TC Electronic. The code is based on a set of parameters you must know before you order the option:

- Serial number of the M5000 frame.
- A four character reference code (only for temporary option installation)

When you want to order an option, either temporary or permanent, you need to supply your dealer with these parameters. You will find them as follows:

SERIAL NUMBER

1. Switch off the M5000.
2. Press the BYPASS-button (**18**) and switch the power on again while pressing the BYPASS-button. Hold it until the SETUP UTILITY-menu appears on the display.
3. Turn the PROGRAM-dial until you find the **SERIAL #** and then press DO.

As you can see on this page, you will find not only the serial number but also the BIOS version and the size of the flash EPROM.

REF. CODE

This reference code is also needed for generating the specific license code and subcode. Follow the procedure below:

4. If you still are in the Setup Utility menu - switch the M5000 off and on again.
5. Press the UTILITY-button (**17**) and turn the PROGRAM-button to the CONFIG menu.
6. Use soft dial A to find the wanted option.
7. When you have found the wanted option press DO once.

OPTION	TIME	LEVEL
(Option name)	--	off

This display shows the status of the selected option.

8. Press DO again;

LICENSE # (XXXX) :	00000000000000000000
CURSOR CHAR	NUM

9. The 4 character reference code for this option is found within the brackets, shown above as (XXXX). This code is only needed when ordering the temporary option with time limit.

Based on the parameters received from you by following the above procedure, a special license code and subcode, which are unique for your frame, are generated at TC Electronic headoffice in Denmark. This is the procedure regardless if you are ordering a temporary or permanent option.

INSTALLING THE OPTION

10. When you have received the 20 character license code and the 8 character subcode, you can follow the steps 5 to 8 above, which should get you to the page where you can dial in the 20 character license code. It is of vital importance when selecting you select the option in step 7 that you select the same option as you got the 4 character reference code from !
11. Use softdial A, B and C as you dial the code.
12. When the license code is dialed please check once again that the characters are dialed correctly.
13. Press DO and dial the subcode. Again double check the dialed characters.
14. Press DO and **switch the M5000 off and then on**. If all codes were dialed correctly and the option selected is correct the option is now installed. **If not:**

The code is illegal Press DO to continue

If this happens then start from step 10 again. Make sure that you select the right option according to the reference code you gave to your dealer.

CHECK THE OPTION INSTALLED

You can check your installation by following procedure:

15. Press the UTILITY-button (**17**) and turn the PROGRAM-button to the CONFIG menu.
16. Scroll trough the options by turning soft dial A. If you want to check one of the options simply press DO. If LEVEL is '**off**' the selected option is not installed. 'TIME' shows the time limit and counts down in hours and minutes unless it is a permanently installed option. The time limit will in that case be 'forever'. When the time is up you will get a warning and at the next power up the temporary option is gone. Press UNDO to go back to the CONFIGURATION MENU.

You have the facility to try the newest options within a certain time limit (normally 100 hours). First of all you need to install the new application software which is described in the 'SOFTWARE INSTALLATION' module in this section.

The option you want to install is in fact already in the M5000. However, a 20 character license code and an 8 character subcode are needed to access the option. This is done by writing a code in the M5000 generated only by TC Electronic. The code is based on a set of parameters you must know before you order the option:

- Serial number of the M5000 frame.
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When you want to order an option, either temporary or permanent, you need to supply your dealer with these parameters. You will find them as follows:

SERIAL NUMBER

1. Switch off the M5000.
2. Press the BYPASS-button (**18**) and switch the power on again while pressing the BYPASS-button. Hold it until the SETUP UTILITY-menu appears on the display.
3. Turn the PROGRAM-dial until you find the **SERIAL #** and then press DO.

As you can see on this page, you will find not only the serial number but also the BIOS version and the size of the flash EPROM.

REF. CODE

This reference code is also needed for generating the specific license code and subcode. Follow the procedure below:

4. If you still are in the Setup Utility menu - switch the M5000 off and on again.
5. Press the UTILITY-button (**17**) and turn the PROGRAM-button to the CONFIG menu.
6. Use soft dial A to find the wanted option.
7. When you have found the wanted option press DO once.

OPTION	TIME	LEVEL
(Option name)	--	off

This display shows the status of the selected option.

8. Press DO again;

LICENSE # (XXXX) :	00000000000000000000
CURSOR CHAR	NUM

9. The 4 character reference code for this option is found within the brackets, shown above as (XXXX). This code is only needed when ordering the temporary option with time limit.

Based on the parameters received from you by following the above procedure, a special license code and subcode, which are unique for your frame, are generated at TC Electronic headoffice in Denmark. This is the procedure regardless if you are ordering a temporary or permanent option.

INSTALLING THE OPTION

10. When you have received the 20 character license code and the 8 character subcode, you can follow the steps 5 to 8 above, which should get you to the page where you can dial in the 20 character license code. It is of vital importance when selecting you select the option in step 7 that you select the same option as you got the 4 character reference code from !
11. Use softdial A, B and C as you dial the code.
12. When the license code is dialed please check once again that the characters are dialed correctly.
13. Press DO and dial the subcode. Again double check the dialed characters.
14. Press DO and **switch the M5000 off and then on**. If all codes were dialed correctly and the option selected is correct the option is now installed. **If not:**

The code is illegal Press DO to continue

If this happens then start from step 10 again. Make sure that you select the right option according to the reference code you gave to your dealer.

CHECK THE OPTION INSTALLED

You can check your installation by following procedure:

15. Press the UTILITY-button (**17**) and turn the PROGRAM-button to the CONFIG menu.
16. Scroll trough the options by turning soft dial A. If you want to check one of the options simply press DO. If LEVEL is '**off**' the selected option is not installed. 'TIME' shows the time limit and counts down in hours and minutes unless it is a permanently installed option. The time limit will in that case be 'forever'. When the time is up you will get a warning and at the next power up the temporary option is gone. Press UNDO to go back to the CONFIGURATION MENU.

Before you can use your purchased option sampler, the SIMM-modules of dynamic ram must be mounted. You can buy SIMM-modules yourself in a normal computer store (see type listings below).

START OF INSTALLATION:

1. Switch of the machine and remove the power cord.
2. Remove the DUM-1 option plate(s) or DSP-module by loosening the 2 screws.

As the M5000 modules are static sensitive devices, certain precautions should be taken to protect them from damage during storage and handling. Please refer also to CON-FIGURATION section, HARDWARE INSTALLATION and MODULE HANDLING.

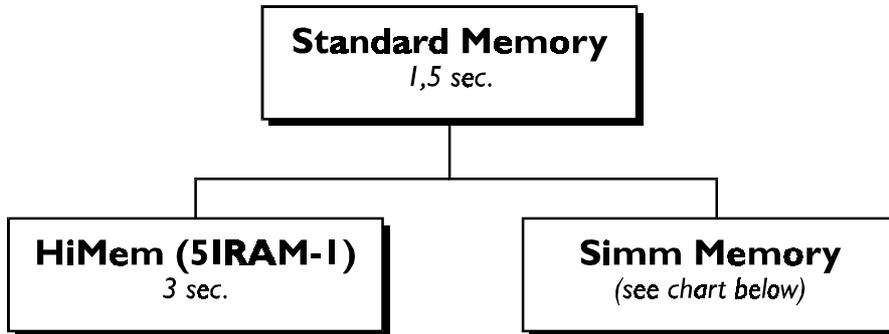
GENERAL HANDLING

When inserting or removing any modules, avoid touching the circuit board by handling only the rear panel of the module. Modules should always be placed in either an M5000 or in an electrostatic shielding bag. To minimize the static potentials that can cause damage to the electronic circuits you should observe precautionary grounding techniques such as touching a grounded M5000 Audio Frame immediately before inserting modules.

REMOVING THE MODULES

Before removing any module from your M5000, switch off the power and unplug the mains power cable. Unplug all other connections from the module before unscrewing the two screws securing the module in the Mainframe.

SELECTION OF THE SIMM PACK MODULES



M5000 Display	Code option	seconds 44.1 (48) KHz	Resolution	No of SIMM's	Type of SIMM's
SAMPLER	5SAMP-3	23.7 (21.8)	16 bit	2	1 MByte x 8 bit
		23.7 (21.8)	18 bit	2	1 MByte x 9 bit
		23.7 (21.8)	24 bit	3	1 MByte x 8 bit
		23.7 (21.8)	24 bit	3	1 MByte x 9 bit
		95.0 (87.3)	16 bit	2	4 MByte x 8 bit
		95.0 (87.3)	18 bit	2	4 MByte x 9 bit
		95.0 (87.3)	24 bit	3	4 MByte x 8 bit
		95.0 (87.3)	24 bit	3	4 MByte x 9 bit
		380.4 (349.2)	16 bit	2	16 MByte x 8 bit
		380.4 (349.2)	18 bit	2	16 MByte x 9 bit
		380.4 (349.2)	24 bit	3	16 MByte x 8 bit
		380.4 (349.2)	24 bit	3	16 MByte x 9 bit

(the sampling times mentioned are max. mono sampling times)

Suggestions to SIMM brands:

- SAMSUNG e.g. KMM594000B-7, 4M x 9 SIMM DRAM Memory Module
- HITACHI
- TEXAS INSTRUMENTS
- TOSHIBA

Speed: fast page mode, 70 ns.

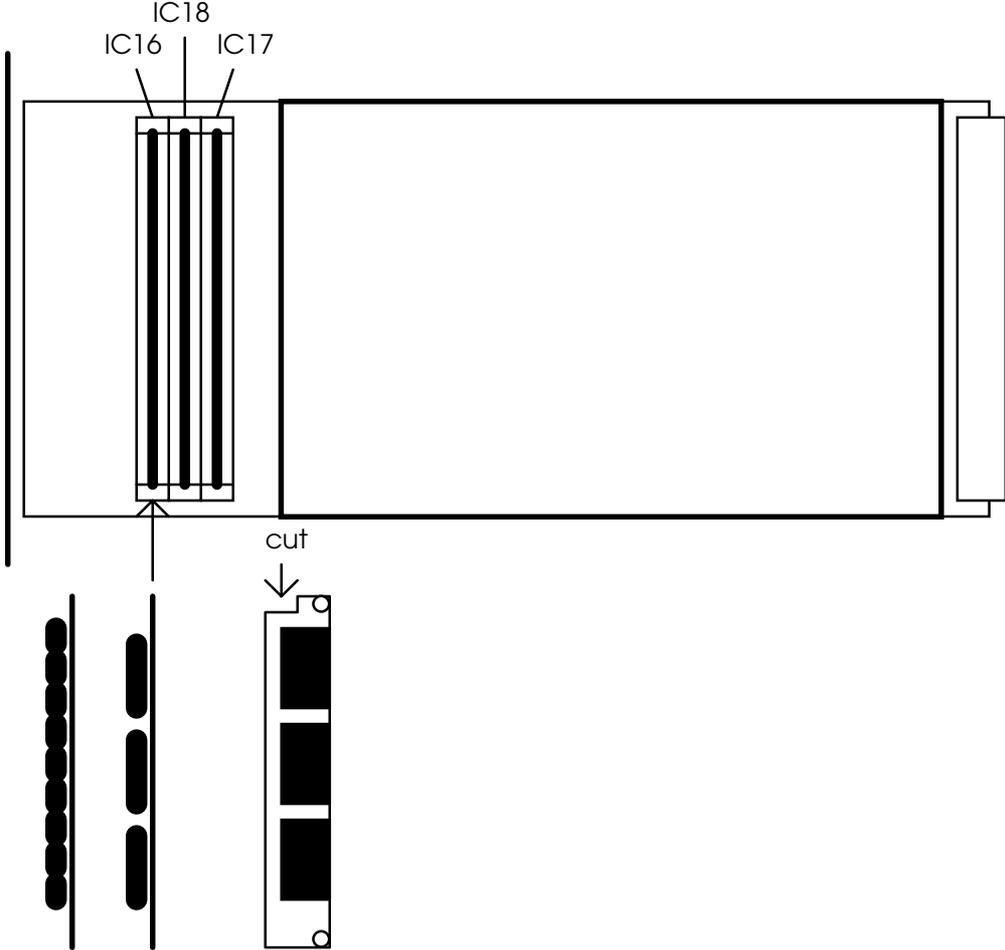
INSTALLING SIMM PACK'S ON THE DSP-MODULE

On the DSP-card between the back plate and the sub boards there are 3 sockets named : IC16, IC17 and IC18.

For installing 16/18 bit sampling, you have to place 2 modules in IC16 and IC18, the sockets nearest to the back plate.

For installing 24-bit sampling, you have to place 3 modules in all three socket positions.

The SIMM-modules can only be properly inserted one way, that is with the IC's on the SIMM-modules facing the back plate. There is an indentation (cut) on the SIMM-modules which must be placed correspondingly with the IC16-IC18-IC17 name reference markings on the board.



MOUNTING MODULES

Before mounting the modules in your M5000, switch off the power and unplug the mains power cable. Remove the dummy-panel or original module from the slot where you want to install the module. The module should then be mounted directly in the M5000 Audio Mainframe by handling the rear panel of the module only. Avoid touching any components on the PCB-Board.

3. Insert the module after the diagram below (fig. 1).

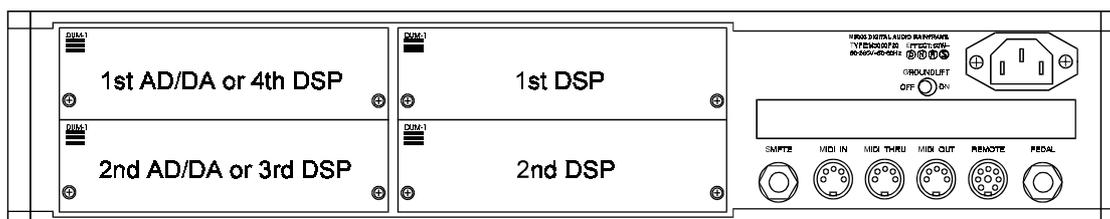


fig. 1

AD/DA cards are always mounted as far as possible from the power supply !

The module cards will fit in the module guides inside the M5000 frame. **It is important that the modules are mounted correctly in these blue guides** to ensure proper connection to the buss. It is recommended that you use a powerful light source in order for you to see properly inside the frame. Improper connection may cause serious damage to the modules.

4. Fasten the module with the two screws and connect the cables.
5. The module cards will be initialized during the next power up.
6. If there are problems e.g., the cards are not recognized by the M5000 frame, please check the installation once again and the address settings (refer to **HARDWARE INSTALLATION** text module in this section).

In Appendix C you can find a self test procedure to see if the M5000 has found the cards at the proper addresses and if the M5000 is working alright.

7. Press the **UTILITY** button on the front panel and turn the **PROGRAM** dial to the **CONFIG** menu. The 'dram=xxxxxx' will tell you if the **SIMM** packs are installed properly. If 'dram=none' is shown then the M5000 hasn't found the **SIMM** packs.

June 26th, 1998
page 1 of 2**Release of 3.52 software for all M5000 owners**

M5000 prior to serial number 283.294: AP1-V352.M5K
M5000X prior to serial number 290.238 2.0: AP1-V352.M5K
M5000 with serial numbers higher than 283.294: AP1-V352.M50
M5000X with serial numbers higher than 290.238 2.0: AP1-V352.M50

Why two different M5000 softwares?

The host processor we were using in the M5000 was becoming unavailable. In order to maintain our service stock of these devices, we even had to stop selling M5000s in the end of 1997.

The new host requires a different but functionally identical software, hence two versions with the same number. In order to prepare all machines for future developments, unfortunately this update has taken some time.

Important information before you update

Please be sure to back-up your presets on a Floppy or to a sequencer before loading the new software.

To maintain preset compatibility between machines, a new preset structure is now implemented.

3.52 machines will read all presets from 1.15 upwards, but 3.52 presets are not readable on M5000s running software prior to 3.50.

When you power on your M5000 after updating, all the internal presets will automatically be changed to 3.5 format, so you have to save the presets before updating to keep the old format also.

New functions and bug fixes (Users updating from version 2.00)**Multistage Phaser Algorithm**

Deep phasing effects have once again become popular, and TC Electronics now respond to several requests of having a monster phaser available on the M5000.

The acclaimed TC XII phaser pedal has been digitized and made stereo, yet maintaining mono compatibility. The swept filter may emulate a structure from 4th to 12th order, and new modulation curves and principles have been added.

Core 2 Reverb Algorithm

When it comes to reverb, it's an advantage to have many different variations available. Some M5 algorithms are focused on effect, others on the shape and characteristic reflections of a specific room.

The new Core2 algorithm is mainly designed with natural rooms and a unidirectional source in mind. The response is dense and smooth without being chorused.

One of the important features about the M5000 reverbs are their true stereo structure. By using extra processing power, the left and right outputs are uncorrelated thereby ensuring mono compatibility. This crucial feature is of course not sacrificed on the new Core2.

Disk to Wizard: If you've also got other TC equipment like M2000, Finalizer, Finalizer Plus, DBMax, Fireworx or G-Force, software may now be transferred to a PCMCIA card from a Floppy disk via your M5000.

The function is found in the Utility menu under File.

Toolbox: The 0.1dB level drop using Toolbox by-pass has been corrected. Please note, that the Toolbox Delay and Lowcut filter are active even when bypassed to avoid glitches. In the Toolbox, turning on and off the Eq bands now no longer produce clicks or pops.

June 26th, 1998
page 2 of 2

New functions and bug fixes (Users updating from version 3.51)

Disk to Wizard: If you've also got other TC equipment like M2000, Finalizer, Finalizer Plus, DBMax, Fireworx or G-Force, software may now be transferred to a PCMCIA card from a Floppy disk via your M5000. The function is found in the Utility menu under File.

Toolbox: The 0.1dB level drop using Toolbox by-pass has been corrected. Please note, that the Toolbox Delay and Lowcut filter are active even when bypassed to avoid glitches.

In the Toolbox, turning on and off the Eq bands no longer produces clicks or pops.

When using Toolbox Eq or Parametric Eq, sometimes wrong values were showed on the display.

Core2 Reverb: Improved presets.

Phaser Algorithm: Improved stereo image.

User Interface: Accelerators on paramater dials have been improved.

MD2: A bug in the algorithm caused changes of settings when exiting certain pages.

MIDI: The MIDI map sometimes was not preserved during power down.

Loading the software

1. Back-up old presets to a disk by pressing Program. Use the "RAM to File" and "File Save Disk" functions.
2. Hold the BYPASS button while you power on the M5000. This gives access to the Setup Utility Menu.
3. Insert the Floppy disk and press DO to load it.
4. Select the new file and press DO twice.
5. After a minute or so the display will read "Keep Previous Settings?"
6. Press UNDO and the new software is ready.

Loading new software from the TC website: www.tcelectronic.com

The latest M5000 software is always available from our website. Mac-users, remember to use a PC formatted disk. When formatting a disk, the friendly Mac will let you choose which format to use.

1. Back-up old presets to a disk by pressing Program. Use the "RAM to File" and "File Save Disk" functions.
2. Check the serial number on your M5000. The serial number is placed on the back panel.
3. Select the correct file for your M5000 at the TC website by use of the tables with serial number areas.
4. Download the file. Don't rename it. The M5000 needs the .M5K or .M50 extension to recognize it.
5. If you have downloaded the zipped version, un-zip it now.
6. Copy the file to a PC formatted floppy disk and insert it in the M5000.
7. Hold the BYPASS button while you power on the M5000. This gives access to the Setup Utility Menu.
8. Insert the Floppy disk and press DO to load it.
9. Select the new file and press DO twice.
10. After a minute or so the display will read "Keep Previous Settings?"
11. Press UNDO and the new software is ready.

AMBIENCE	REVCORE2
100 TlfBooth	132 InstrRoom
101 TileBath	133 DarkRoom
102 LongTube	134 DarkHall
103 Festival	135 PercRoom
104 NextDoor	136 LiveRoom
105 Garage	137 Chuch 3
106 Van1	138 WoodRoom
107 Van2	PHASER-1
108 SlugBug	139 Phase 1
109 Wreck	140 Trash
110 Studio1	141 Sgt.#1
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TAPFAC	143 Deep #1
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114 MultiTap	DYNAMIC1
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119 TrueRoom	206 RockLim1
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123 The Shop	210 EasyExp1
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125 CORERoom	212 RecComp1
126 DrewRoom	213 CDMaster
127 X&Y Mics	TOOLBOX
128 Closet	214 Neutral
129 NewBooth	
130 Stage	
131 At Home	

Environmental - Telephone booth
Environmental - Bathroom with tiles
Environmental - Sewer tube
Environmental - Big Rocknroll festival
Environmental - The neighbors having a party
Environmental
Environmental - frontseat sound '67VWvan
Environmental - luggage compartment with no interior '67VWvan
Environmental - VolksWagen
Environmental - Bad car stereo
General purpose ambience
General purpose ambience

REVCORE-1 algorithm programs



PAGE	1A	1B	2A	2B	2C	2D	2E	2F	3A	3B	3C	3D	3E	4A	4B	5A	5B	5C	5C	
No	Name	Notes	Mix In	Out	Decay	xLow	xHigh	InitLev	RevLev	Lm-Xovr	Mh-Xovr	Shape	Xsize	PreDly	RevFeed	Hicut	Att	Spread	DiffTyp	R-Width
#	8 CHAR.		%	dB	s			dB	dB	Hz	Hz	x	ms	ms	ms	Hz	dB			%
119	TrueRoom		34	0	0.5	0.61	0.47	-14	0	800	4	Hall	0.25	0	0	3.15	-31.5	1	1	100
120	HomeRoom		24	0	0.7	0.92	0.75	-10.5	0	125	3.15	Prism	0.4	0	0	4	-31.5	1	1	100
121	WoodChmb		39	0	1.4	0.7	0.77	-10.5	0	100	3.15	Fan	0.4	7.3	0	2.5	-23	1	1	63
122	Goldfoil		36	0	2.2	1.09	1.18	off	0	315	10	Fan	0.4	7.3	0	8	-23	1	1	100
123	The Shop		31	0	2.2	0.59	0.33	-4.5	0	1.25	2.5	H.shoe	0.25	41.5	0	4	-10.5	1	1	100
124	Fridge		33	0	0.4	0.83	0.27	-11.5	0	800	3.15	Prism	0.4	0	0	4	-31.5	1	1	84
125	COREroom		38	0	0.4	0.79	0.64	-11	0	500	5	Small	0.5	0	0	1.6	-25	1	1	100
126	DrewRoom		44	0	0.9	0.01	0.57	-8	0	32	3.15	Small	0.5	0	0	4	-14	1	1	93
127	X&Y Mics		27	0	0.3	0.59	0.07	-5	0	250	1.6	H.shoe	0.125	0	0	Flat	-37.5	1	1	100
128	Closet		39	0	0.4	0.88	0.13	-7.5	0	800	2	Small	0.316	1.1	0	4	-31.5	1	1	84
129	NewBooth		41	0	0.5	0.66	0.47	-16	0	500	2	Prism	0.16	0	0	5	-34	1	1	100
130	Stage		34	0	2.2	0.82	0.5	-9	0	800	2.5	H.shoe	0.63	19.3	7.1	2.5	-39	1	1	100
131	At Home		37	0	0.5	0.59	0.28	-18	0	160	4	Fan	0.316	1.1	0	2.5	-31	1	0	49

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REVERB-1 algorithm programs



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No	Name	Mix	In	Decay	xLow	xHigh	Diffuse	Shape	xSize	PreDly	RevFeed	HiCut	Att	Lo-Xovr	Hi-Xovr	InitLev	RevLev	RWidth	I-XFeed
#	8 CHAR.	%	dB	s					x	ms	ms	Hz	dB	Hz	Hz	dB	dB		
1	Church 1	25	0	0	4.2	0.7	0.18	11	fan	1.25	30	0	-30	125	8K	0	0	100	on
2	Church 2	20	0	0	2	1.2	0.6	11	h.shoe	0.63	27	50	0	315	8K	-4	0	100	on
3	480 Hall	20	0	0	3.5	1	0.45	13	fan	0.8	30	20	-1.5	250	8K	0	-2	100	on
4	VocalDry	30	0	0	0.6	1	0.4	6	hall	0.4	18	15	-6	250	8K	0	-12.5	75	on
5	VocalWet	25	0	0	1.2	0.6	0.6	11	h.shoe	0.63	0	50	-8	500	6K3	-4	0	100	on
6	ManInBox	40	0	0	0.3	0.2	1.5	6	hall	0.2	20	26	-6	250	8K	0	-12	70	on
7	Locker	25	0	0	0.8	0.9	0.8	6	prism	0.4	4	0	-24	3K15	3K15	0	-6	70	on
8	DryHouse	25	0	0	0.5	1	0.4	6	hall	0.63	22	0	-6	250	8K	0	-10	20	on
9	WetHouse	30	0	0	1.4	1	0.4	8	hall	0.63	22	0	-6	250	8K	0	-10	20	on
10	Stage	20	0	0	2	1	0.38	7	hall	0.63	22	75	-20	250	8K	0	0	85	on
11	Rattle	25	0	0	1.4	1	0.4	1	h.shoe	2.5	18	18	-6	250	8K	0	-10	100	on
12	ShortCut	30	0	0	0.3	1	1	6	h.shoe	0.63	7	40	-25	250	5K	-5	0	70	on
13	SlapHall	30	0	0	1	1	0.45	6	hall	0.316	120	60	-6	250	8K	0	0	90	on
14	Ugly 1	30	0	0	1	0.65	0.45	5	prism	1.25	10	30	-30	800	800	0	-3	70	on
15	Ugly 2	30	0	0	1	1	0.1	5	hall	1.25	10	30	-30	800	8K	0	-3	60	on

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Created by Thomas Olesen, Tom Andersen and Ivar Iversen in FEEDBACK Studio 1+2, Aarhus, Denmark.

PITCH-1 algorithm programs

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No	Name	Notes	Mix %	Inlev dB	Outlev dB	Pitch-1 %	Fine-1 %	Pitch-2 %	Fine-2 %	Level-1 %	Pan-1 %	Level-2 %	Pan-2 %	Hicut-1 Hz	Att-1 dB	Hicut-2 Hz	Att-2 dB	FB-1 %	FB-2 %	Xfb1>2 %	Xfb2>1 %	Delay-1 mS	Delay-2 mS	Dgspeed	Polyspd	Polydly	Dgfltt	
#	8	CHAR.																										KHZ
73	Slapitch	Slap Guitar	30	0	0	0	-10	0	0	0	50 R	0	50 L	10k	-30	10k	-30	50	50	0	100	10	84	0.50	50	18	2	
74	PowerOct	Octave below	45	0	0	-12	-1200			0	50 L	0	50 R	flat	0	flat	0	0	0	0	0	0	0	0.10	20	18	2	
75	Valley	Detuned room	20	0	0	0	-10	0	10	0	50 L	0	50 R	10	-40	10	-40	50	50	0	100	40	60	0.50	50	18	2	
76	Climb	Pitch climbing up	50	0	0	7	700	3	300	0	50 R	0	50 L	flat	0	flat	0	30	30	20	20	310	155	0.28	50	18	2	
77	Barbshop	Minor 3rd blw, 5th abv.	60	0	0	-3	-300	7	700	0	Center	0	Center	flat	0	flat	0	0	0	0	0	0	0	0.28	50	18	2	
78	Fifths	5th above and below	50	0	0	6	690	7	710	0	50 R	0	50 L	flat	0	flat	0	0	0	0	0	0	0	0.28	50	18	2	
79	Octave+	Octave above	35	0	0	11	1190	12	1200	0	50 R	0	50 L	flat	0	flat	0	0	0	0	0	0	0	0.28	50	18	2	
80	Chord	Major triad chord	60	0	0	5	500	9	900	0	10 R	0	10 L	flat	0	flat	0	0	0	0	0	0	0	0.28	50	18	2	
81	Horror	Spacey pitch	60	0	0	-1	-100	1	100	0	50 L	0	50 R	flat	0	flat	0	50	50	30	50	100	140	0.28	50	18	2	
82	Steel	Metallic pitch	50	0	0	11	1190	12	1200	0	50 L	0	50 R	flat	0	flat	0	20	20	50	60	50	50	0.28	50	18	2	
83	WideBass	Chorus for bass	40	0	0	0	8	0	-8	0	center	0	center	flat	0	flat	0	0	0	0	0	0	0	0.10	20	12	2	
84	VocalFml	Pitch for female vocal	67	0	0	0	14	0	-13	0	50 L	0	50 R	6.3k	-6	6.3k	-3	0	0	0	0	0	0	0.20	20	10	2	
85	Vocals	General vocal pitch	65	0	0	-3	-300	0	-18	-10	20 R	-6.5	20 L	6.3k	-20	6.3k	-20	0	0	0	0	0	0	0.20	20	18	2	

SAMPLE-1 algorithm programs.



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No	Name	Notes	Mix %	Inlev dB	Outlev dB	Sample	Action Mode	Counter Sek.	Status	Start Sek.	Fine mS	End Sek.	Reverse On/Off	Recmode Mode	Fine mS	End Sek.	Level dB	Pan	Fadein sek.	Fadeovr sek.	Trigger	Triglev dB	Deadbnd dB	Retrig Sek.		
99	Sample	Standard sampler	100	0	0	sample	None	0	ready?	0	0	0	0	mono	0	0	0	center	0	0	0.01	manual	-25	-3	0.1	

TAPFAC Algorithm presets



Preset: 112	Use this preset for:
MoneyBox	

1	MIX	INLEV	OUTLEV
	50	0	0

2	Scale (%)	PreDly (ms)	Width (%)	LastTap
	20	0	100	18

3	Tap	Delay (ms)	Level (%)	Pan (±10)
	1	3.4	32	8L
	2	15.5	71	4R
	3	31.5	63	7L
	4	33.9	56	5R
	5	38.1	50	10L
	6	59.2	45	10R
	7	55.6	40	4R
	8	83.4	35	10L
	9	84.5	31	10R
	10	93.6	28	5R
	11	134	25	7L
	12	152.9	22	5R
	13	205.2	20	6L
	14	248.7	18	5R
	15	293.2	16	6L
	16	400.3	14	7R
	17	493.4	12	6L
	18	615.3	11	7R

4	LoCut (Hz)	Att (dB)	HiCut (Hz)	Att (dB)
	20	0	6.3K	-40

5	Speed (Hz)	Depth (%)
	0.2	0

Preset: 113	Use this preset for:
Atmosph1	

1	MIX	INLEV	OUTLEV
	50	0	0

2	Scale (%)	PreDly (ms)	Width (%)	LastTap
	25	0	100	18

3	Tap	Delay (ms)	Level (%)	Pan (±10)
	1	5.5	32	8L
	2	29.3	71	4R
	3	44.5	63	7L
	4	63.3	56	6R
	5	78.5	50	10L
	6	91.6	45	10R
	7	106.8	40	4R
	8	140.6	35	10L
	9	179.3	31	10R
	10	210	28	6R
	11	253.9	25	7L
	12	297.5	22	5R
	13	327.8	20	6L
	14	385.8	18	5R
	15	430.8	16	6L
	16	479.9	14	7R
	17	555.4	12	6L
	18	621.6	11	7R

4	LoCut (Hz)	Att (dB)	HiCut (Hz)	Att (dB)
	20	0	2.5k	-20

5	Speed (Hz)	Depth (%)
	0.2	0

Preset: 114	Use this preset for:
MultiTap	

1	MIX	INLEV	OUTLEV
	50	0	0

2	Scale (%)	PreDly (ms)	Width (%)	LastTap
	100	0	100	18

3	Tap	Delay (ms)	Level (%)	Pan (±10)
	1	22.5	32	8L
	2	73.2	71	4R
	3	119.9	63	7L
	4	162.8	56	6R
	5	220.4	50	10L
	6	263.2	45	10R
	7	299.7	40	4R
	8	325.7	35	10L
	9	383.1	31	10R
	10	419.4	28	6R
	11	427.1	25	7L
	12	480.1	22	5R
	13	489.7	19	6L
	14	519	18	5R
	15	546.3	16	6L
	16	585.6	14	7R
	17	602.7	12	6L
	18	623	11	7R

4	LoCut (Hz)	Att (dB)	HiCut (Hz)	Att (dB)
	20	0	2.5k	-20

5	Speed (Hz)	Depth (%)
	0.2	0



TAPFAC Algorithm presets



Preset: 115
BeatBox1

Use this preset for:

1	MIX	INLEV	OUTLEV
	50	0	0

2	Scale (%)	PreDly (ms)	Width (%)	LastTap
	100	0	100	14

3	Tap	Delay (ms)	Level (%)	Pan (±10)
	1	100	70	10R
	2	100.7	70	10R
	3	100	63	10L
	4	200.6	56	10L
	5	300	50	10R
	6	300.8	45	10R
	7	400	40	10L
	8	400.5	35	10L
	9	400	31	10R
	10	400.7	28	10R
	11	500	22	10L
	12	500.4	22	10L
	13	600	19	10R
	14	600.5	19	10R
	15			
	16			
	17			
	18			

4	LoCut (Hz)	Att (dB)	HiCut (Hz)	Att (dB)
	20	0	2.5k	0

5	Speed (Hz)	Depth (%)
		0

Preset: 116
BeatBox2

Use this preset for:

1	MIX	INLEV	OUTLEV
	25	0	0

2	Scale (%)	PreDly (ms)	Width (%)	LastTap
	77	0	100	4

3	Tap	Delay (ms)	Level (%)	Pan (±10)
	1	150	50	10L
	2	300	50	2L
	3	450	50	3R
	4	600	100	0
	5			
	6			
	7			
	8			
	9			
	10			
	11			
	12			
	13			
	14			
	15			
	16			
	17			
	18			

4	LoCut (Hz)	Att (dB)	HiCut (Hz)	Att (dB)
	20	0	2.5k	0

5	Speed (Hz)	Depth (%)
		0

Preset: 117
FlamBeat

Use this preset for:

1	MIX	INLEV	OUTLEV
	34	0	0

2	Scale (%)	PreDly (ms)	Width (%)	LastTap
	80	0	100	5

3	Tap	Delay (ms)	Level (%)	Pan (±10)
	1	150	80	10L
	2	300	100	10R
	3	465	100	10L
	4	35	100	10R
	5	35	100	0
	6			
	7			
	8			
	9			
	10			
	11			
	12			
	13			
	14			
	15			
	16			
	17			
	18			

4	LoCut (Hz)	Att (dB)	HiCut (Hz)	Att (dB)
	20	0	2.5k	0

5	Speed (Hz)	Depth (%)
	0.2	50



DYNAMIC1 Algorithm

Preset: 200 Use this preset for:

1BandCom

1	MIX	INLEV	OUTLEV	Balance
	100%	0.0dB	0.0dB	center

2	LowCut	Lo-Xovr	Hi-Xovr	SoftClip
	2.0Hz	low-off	mid-off	on

3	LEVELS	low	mid	high
	Bnd-Lev	0.0dB	0.0dB	0.0dB
	0dB ref	-8.0dB	-8.0dB	-8.0dB
	Meters	5dB	5dB	5dB

4	COMPRES	low	mid	high
	Threshold	-4.0dB	-4.0dB	-4.0dB
	Ratio	2.0>1	2.0>1	2.0>1
	Gain	2.0dB	2.0dB	2.0dB
	Attack	20ms	20ms	20ms
	Release	500ms	500ms	500ms
	FeedFwd	10.0ms	10.0ms	10.0ms
	Crest	RMS	RMS	RMS

5	LIMITER	low	mid	high
	Threshold	0.0dB	0.0dB	0.0dB
	Ratio	infin>1	infin>1	infin>1
	Attack	1.4ms	1.4ms	1.4ms
	Release	1.4s	1.4s	1.0s
	FeedFwd	1.0ms	1.0ms	1.0ms

6	EXPAND	low	mid	high
	Threshold	-40.0dB	-40.0dB	-40.0dB
	Ratio	off	off	off
	Attack	1.0ms	1.0ms	0.3ms
	Release	1.0s	1.0s	1.0s
	Range	-30.0dB	-30.0dB	-30.0dB

7	Par-Lnk	Norm-Dly
	on	10.0ms

Preset: 201 Use this preset for:

2BandCom

1	MIX	INLEV	OUTLEV	Balance
	100%	0.0dB	0.0dB	center

2	LowCut	Lo-Xovr	Hi-Xovr	SoftClip
	2.0Hz	low-off	400Hz	on

3	LEVELS	low	mid	high
	Bnd-Lev	0.0dB	-0.5dB	-2.0dB
	0dB ref	-8.0dB	-8.0dB	-8.0dB
	Meters	5dB	5dB	5dB

4	COMPRES	low	mid	high
	Threshold	0.0dB	-7.5dB	-6.5dB
	Ratio	2.0>1	2.0>1	2.0>1
	Gain	0.0dB	3.7dB	3.2dB
	Attack	20ms	20ms	20ms
	Release	500ms	500ms	500ms
	FeedFwd	10.0ms	10.0ms	10.0ms
	Crest	RMS	RMS	RMS

5	LIMITER	low	mid	high
	Threshold	0.0dB	0.0dB	0.0dB
	Ratio	infin>1	infin>1	infin>1
	Attack	1.4ms	1.4ms	1.4ms
	Release	1.4s	1.4s	1.0s
	FeedFwd	1.0ms	1.0ms	1.0ms

6	EXPAND	low	mid	high
	Threshold	-40.0dB	-40.0dB	-40.0dB
	Ratio	off	off	off
	Attack	1.0ms	1.0ms	0.3ms
	Release	1.0s	1.0s	1.0s
	Range	-30.0dB	-30.0dB	-30.0dB

7	Par-Lnk	Norm-Dly
	off	10.0ms



Preset: 202 Use this preset for:

3BandCom

1	MIX	INLEV	OUTLEV	Balance
	100%	0.0dB	0.0dB	center

2	LowCut	Lo-Xovr	Hi-Xovr	SoftClip
	2.0Hz	100Hz	3.15KHz	on

3	LEVELS	low	mid	high
	Bnd-Lev	-2.0dB	-1.0dB	-1.0dB
	0dB ref	-8.0dB	-8.0dB	-8.0dB
	Meters	5dB	5dB	5dB

4	COMPRES	low	mid	high
	Threshold	-8.5dB	-2.5dB	-10.5dB
	Ratio	4.0>1	2.5>1	2.0>1
	Gain	6.3dB	1.5dB	5.2dB
	Attack	30ms	20ms	20ms
	Release	300ms	500ms	700ms
	FeedFwd	10.0ms	10.0ms	10.0ms
	Crest	RMS	RMS	RMS

5	LIMITER	low	mid	high
	Threshold	0.0dB	0.0dB	0.0dB
	Ratio	infin>1	infin>1	infin>1
	Attack	1.4ms	1.4ms	1.4ms
	Release	1.4s	1.4s	1.0s
	FeedFwd	1.0ms	1.0ms	1.0ms

6	EXPAND	low	mid	high
	Threshold	-40.0dB	-40.0dB	-40.0dB
	Ratio	off	off	off
	Attack	1.0ms	1.0ms	0.3ms
	Release	1.0s	1.0s	1.0s
	Range	-30.0dB	-30.0dB	-30.0dB

7	Par-Lnk	Norm-Dly
	off	10.0ms



DYNAMIC1 Algorithm

Preset: 203
TapeSim1

Use this preset for:

1	MIX	INLEV	OUTLEV	Balance
	100%	0dB	0dB	center

2	LowCut	Lo-Xovr	Hi-Xovr	SoftClip
	2.0hz	400Hz	2.5KHz	on

3	LEVELS	low	mid	high
	Bnd-Lev	0dB	-0.5dB	-3.5dB
	0dB ref	0dB	0dB	0dB
	Meters	10dB	10dB	10dB

4	COMPRES	low	mid	high
	Threshold	-14dB	-14dB	-18dB
	Ratio	1.6>1	1.6>1	1.8>1
	Gain	5.2dB	5.2dB	7.9dB
	Attack	3.0ms	2.0ms	1.4ms
	Release	1s	700ms	300ms
	FeedFwd	4.0ms	3.0ms	2.5ms
	Crest	12dB	12dB	12dB

5	LIMITER	low	mid	high
	Threshold	-1.5dB	-2.5dB	-4.0dB
	Ratio	infin>1	infin>1	infin>1
	Attack	100us	100us	50us
	Release	200ms	200ms	200ms
	FeedFwd	3.0ms	1.0ms	1.0ms

6	EXPAND	low	mid	high
	Threshold			
	Ratio	off	off	off
	Attack			
	Release			
	Range			

7	Par-Lnk	Nom-Dly
	off	10ms

Preset: 204
TapeSim2

Use this preset for:

1	MIX	INLEV	OUTLEV	Balance
	100%	0dB	0.0dB	center

2	LowCut	Lo-Xovr	Hi-Xovr	SoftClip
	2.0hz	315Hz	4KHz	on

3	LEVELS	low	mid	high
	Bnd-Lev	-1.0dB	-3.5dB	-8.0dB
	0dB ref	0dB	0dB	0dB
	Meters	20dB	20dB	20dB

4	COMPRES	low	mid	high
	Threshold	-14dB	-14dB	-18dB
	Ratio	3.2>1	3.2>1	3.2>1
	Gain	9.6dB	9.6dB	12.3dB
	Attack	3.0ms	2.0ms	1.4ms
	Release	1.0s	700ms	300ms
	FeedFwd	4ms	3ms	2.5ms
	Crest	12dB	12dB	12dB

5	LIMITER	low	mid	high
	Threshold	-1.0dB	-1.0dB	-6.0dB
	Ratio	infin>1	infin>1	infin>1
	Attack	100us	100us	50us
	Release	300ms	140ms	50ms
	FeedFwd	0.1ms	0.1ms	0.1ms

6	EXPAND	low	mid	high
	Threshold			
	Ratio	off	off	off
	Attack			
	Release			
	Range			

7	Par-Lnk	Nom-Dly
	off	3ms



Preset:

Use this preset for:

1	MIX	INLEV	OUTLEV	Balance

2	LowCut	Lo-Xovr	Hi-Xovr	SoftClip

3	LEVELS	low	mid	high
	Bnd-Lev			
	0dB ref			
	Meters			

4	COMPRES	low	mid	high
	Threshold			
	Ratio			
	Gain			
	Attack			
	Release			
	FeedFwd			
	Crest			

5	LIMITER	low	mid	high
	Threshold			
	Ratio			
	Attack			
	Release			
	FeedFwd			

6	EXPAND	low	mid	high
	Threshold			
	Ratio			
	Attack			
	Release			
	Range			

7	Par-Lnk	Nom-Dly



DYNAMIC1 Algorithm

Preset: 205
Loudness

Use this preset for:

1	MIX	INLEV	OUTLEV	Balance
	100%	0dB	0.0dB	center

2	LowCut	Lo-Xovr	Hi-Xovr	SoftClip
	2.0hz	200Hz	4.0KHz	on

3	LEVELS	low	mid	high
	Bnd-Lev	1.0dB	-1.5dB	1.0dB
	0dB ref	-6dB	-6dB	-6dB
	Meters	10dB	10dB	10dB

4	COMPRES	low	mid	high
	Threshold	-10dB	-15dB	-12.5dB
	Ratio	8.0>1	3.2>1	5.6>1
	Gain	8.7dB	10.3dB	10.2dB
	Attack	2.0ms	2.0ms	2.0ms
	Release	50ms	1.0s	1.0s
	FeedFwd	1.0ms	10ms	10ms
	Crest	rms	peak	peak

5	LIMITER	low	mid	high
	Threshold	-2.0dB	-2.0dB	-2.0dB
	Ratio	infin>1	infin>1	infin>1
	Attack	50us	50us	50us
	Release	200ms	200ms	200ms
	FeedFwd	3.0ms	1.0ms	1.0ms

6	EXPAND	low	mid	high
	Threshold			
	Ratio	off	off	off
	Attack			
	Release			
	Range			

7	Par-Lnk	Nom-Dly
	off	10ms

Preset: 206
RockLim1

Use this preset for:

1	MIX	INLEV	OUTLEV	Balance
	100%	0dB	0dB	center

2	LowCut	Lo-Xovr	Hi-Xovr	SoftClip
	2.0hz	400Hz	3.15KHz	on

3	LEVELS	low	mid	high
	Bnd-Lev	-1.0dB	-3.0dB	-2.5dB
	0dB ref	-6dB	-6dB	-6dB
	Meters	10dB	10dB	10dB

4	COMPRES	low	mid	high
	Threshold	-13.5dB	-12dB	-15dB
	Ratio	32>1	32>1	32>1
	Gain	13.0dB	11.6dB	14.5dB
	Attack	10ms	10ms	10ms
	Release	1.0s	1.0s	1.0s
	FeedFwd	10ms	10ms	10ms
	Crest	12dB	12dB	12dB

5	LIMITER	low	mid	high
	Threshold	-4.5dB	-6.5dB	-8.0dB
	Ratio	infin>1	infin>1	infin>1
	Attack	100us	140us	100us
	Release	50ms	50ms	50ms
	FeedFwd	3.0ms	1.0ms	1.0ms

6	EXPAND	low	mid	high
	Threshold			
	Ratio	off	off	off
	Attack			
	Release			
	Range			

2	Par-Lnk	Nom-Dly
	off	10ms



Preset: 207
Hi-Fi

Use this preset for:

1	MIX	INLEV	OUTLEV	Balance
	100%	0dB	0dB	center

2	LowCut	Lo-Xovr	Hi-Xovr	SoftClip
	2.0Hz	63Hz	2.50KHz	on

3	LEVELS	low	mid	high
	Bnd-Lev	-5.0dB	0.0dB	-3.0dB
	0dB ref	-8.0dB	-8.0dB	-8.0dB
	Meters	5dB	5dB	5dB

4	COMPRES	low	mid	high
	Threshold	-13.0dB	0.0dB	-16.0dB
	Ratio	8.0>1	2.5>1	2.0>1
	Gain	11.3dB	0.0dB	8.0dB
	Attack	30ms	20ms	20ms
	Release	500ms	500ms	700ms
	FeedFwd	10.0ms	10.0ms	10.0ms
	Crest	RMS	RMS	RMS

5	LIMITER	low	mid	high
	Threshold	0.0dB	0.0dB	0.0dB
	Ratio	infin>1	infin>1	infin>1
	Attack	1.4ms	1.4ms	1.4ms
	Release	1.4s	1.4s	1.4s
	FeedFwd	1.0ms	1.0ms	1.0ms

6	EXPAND	low	mid	high
	Threshold	-40.0dB	-40.0dB	-40.0dB
	Ratio	off	off	off
	Attack	1.0ms	1.0ms	0.3ms
	Release	1.0s	1.0s	1.0s
	Range	-30dB	-30dB	-30dB

7	Par-Lnk	Nom-Dly
	off	10.0ms

DYNAMIC1 Algorithm

Preset: 208
Gain

Use this preset for:

1	MIX	INLEV	OUTLEV	Balance
	100%	0.0dB	0.0dB	center

2	LowCut	Lo-Xovr	Hi-Xovr	SoftClip
	2.0Hz	low-off	mid-off	on

3	LEVELS	low	mid	high
	Bnd-Lev	3.0dB	3.0dB	3.0dB
	0dB ref	-8.0dB	-8.0dB	-8.0dB
	Meters	5dB	5dB	5dB

4	COMPRES	low	mid	high
	Threshold	0.0dB	0.0dB	0.0dB
	Ratio	off	off	off
	Gain	0.0dB	0.0dB	0.0dB
	Attack	30ms	20ms	20ms
	Release	500ms	500ms	700ms
	FeedFwd	10.0ms	10.0ms	10.0ms
	Crest	RMS	RMS	RMS

5	LIMITER	low	mid	high
	Threshold	0.0dB	0.0dB	0.0dB
	Ratio	infin>1	infin>1	infin>1
	Attack	1.0ms	1.0ms	1.0ms
	Release	1.4s	1.4s	1.4s
	FeedFwd	1.0ms	1.0ms	1.0ms

6	EXPAND	low	mid	high
	Threshold	-40dB	-40dB	-40dB
	Ratio	off	off	off
	Attack	1.0ms	1.0ms	0.3ms
	Release	1.0s	1.0s	1.0s
	Range	-30.0dB	-30.0dB	-30.0dB

7	Par-Lnk	Norm-Dly
	on	10.0ms

Preset: 209
CompAnd

Use this preset for:

1	MIX	INLEV	OUTLEV	Balance
	100%	0.0dB	0.0dB	center

2	LowCut	Lo-Xovr	Hi-Xovr	SoftClip
	2.0Hz	100Hz	3.15KHz	on

3	LEVELS	low	mid	high
	Bnd-Lev	-2.0dB	-1.0dB	-1.0dB
	0dB ref	-8.0dB	-8.0dB	-8.0dB
	Meters	5dB	5dB	5dB

4	COMPRES	low	mid	high
	Threshold	-8.5dB	-2.5dB	-10.5dB
	Ratio	4.0>1	2.5>1	2.0>1
	Gain	6.3dB	1.5dB	5.2dB
	Attack	30ms	20ms	20ms
	Release	300ms	500ms	700ms
	FeedFwd	10.0ms	10.0ms	10.0ms
	Crest	RMS	RMS	RMS

5	LIMITER	low	mid	high
	Threshold	-1.5dB	-1.5dB	-1.5dB
	Ratio	infin>1	infin>1	infin>1
	Attack	1.4ms	1.4ms	1.4ms
	Release	1.4s	1.4s	1.4s
	FeedFwd	1.0ms	1.0ms	1.0ms

6	EXPAND	low	mid	high
	Threshold	-24.0dB	-24.0dB	-24.0dB
	Ratio	1>3.2	1>3.2	1>3.2
	Attack	0.3ms	0.3ms	0.3ms
	Release	1.0s	1.0s	1.0s
	Range	-40.0dB	-40.0dB	-40.0dB

7	Par-Lnk	Norm-Dly
	off	10.0ms



Preset: 210
EasyExp1

Use this preset for:

1	MIX	INLEV	OUTLEV	Balance
	100%	0.0dB	0.0dB	center

2	LowCut	Lo-Xovr	Hi-Xovr	SoftClip
	2.0Hz	200Hz	1.60KHz	off

3	LEVELS	low	mid	high
	Bnd-Lev	-3.0dB	-3.0dB	-6.0dB
	0dB ref	0.0dB	0.0dB	0.0dB
	Meters	10dB	10dB	10dB

4	COMPRES	low	mid	high
	Threshold	-14.0dB	-14.0dB	-18.0dB
	Ratio	2.0>1	2.0>1	2.5>1
	Gain	7.0dB	7.0dB	10.8dB
	Attack	3.0ms	2.0ms	1.4ms
	Release	1.0s	700ms	300ms
	FeedFwd	3.0ms	3.0ms	2.5ms
	Crest	12dB	12dB	12dB

5	LIMITER	low	mid	high
	Threshold	-1.5dB	-2.5dB	-4.0dB
	Ratio	infin>1	infin>1	infin>1
	Attack	100us	100us	50us
	Release	300ms	140ms	70ms
	FeedFwd	0.2ms	0.2ms	0.2ms

6	EXPAND	low	mid	high
	Threshold	-32.0dB	-38dB	-46dB
	Ratio	1>2.0	1>2.0	1>2.0
	Attack	1.0ms	1.0ms	1.0ms
	Release	300ms	300ms	300ms
	Range	-20dB	-20dB	-20dB

7	Par-Lnk	Norm-Dly
	off	3.0ms



DYNAMIC1 Algorithm

Preset: 211	Use this preset for:
SoftLim	

1	MIX	INLEV	OUTLEV	Balance
	100%	0.0dB	0.0dB	center

2	LowCut	Lo-Xovr	Hi-Xovr	SoftClip
	2.0Hz	low-off	mid-off	on

3	LEVELS	low	mid	high
	Bnd-Lev	4.0dB	4.0dB	4.0dB
	0dB ref	0.0dB	0.0dB	0.0dB
	Meters	10dB	10dB	10dB

4	COMPRES	low	mid	high
	Threshold	0.0dB	0.0dB	0.0dB
	Ratio	off	off	off
	Gain	0.0dB	0.0dB	0.0dB
	Attack	3.0ms	2.0ms	1.4ms
	Release	1.0s	700ms	300ms
	FeedFwd	0.0ms	0.0ms	0.0ms
	Crest	12dB	12dB	12dB

5	LIMITER	low	mid	high
	Threshold	0.0dB	0.0dB	0.0dB
	Ratio	off	off	off
	Attack	100µs	100µs	50µs
	Release	300ms	140ms	70ms
	FeedFwd	0.2ms	0.2ms	0.2ms

6	EXPAND	low	mid	high
	Threshold	-32.0dB	-38.0dB	-46dB
	Ratio	off	off	off
	Attack	1.0ms	1.0ms	1.0ms
	Release	300ms	300ms	300ms
	Range	-20.0dB	-20.0dB	-20.0dB

7	Par-Lnk	Norm-Dly
	on	0.0ms

Preset: 212	Use this preset for:
RecComp 1	

1	MIX	INLEV	OUTLEV	Balance
	100%	0.0dB	0.0dB	center

2	LowCut	Lo-Xovr	Hi-Xovr	SoftClip
	8.0Hz	200Hz	2.00KHz	on

3	LEVELS	low	mid	high
	Bnd-Lev	-3.0dB	-3.5dB	-5.5dB
	0dB ref	0.0dB	0.0dB	0.0dB
	Meters	10dB	10dB	10dB

4	COMPRES	low	mid	high
	Threshold	-14.0dB	-14.0dB	-18.0dB
	Ratio	1.80>1	1.80>1	2.0>1
	Gain	6.2dB	6.2dB	9.0dB
	Attack	2.0ms	1.4ms	1.0ms
	Release	1.0s	700ms	300ms
	FeedFwd	2.0ms	1.4ms	0.5ms
	Crest	10dB	10dB	10dB

5	LIMITER	low	mid	high
	Threshold	-1.5dB	-2.5dB	-4.0dB
	Ratio	infin>1	infin>1	infin>1
	Attack	100µs	100µs	50µs
	Release	300ms	140ms	70ms
	FeedFwd	0.2ms	0.2ms	0.2ms

6	EXPAND	low	mid	high
	Threshold	-64dB	-64dB	-70dB
	Ratio	1>2.0	1>2.0	1>2.0
	Attack	1.0ms	0.7ms	0.5ms
	Release	700ms	700ms	700ms
	Range	-10dB	-10dB	-10dB

7	Par-Lnk	Norm-Dly
	off	0.3ms



Preset:	Use this preset for:

1	MIX	INLEV	OUTLEV	Balance

2	LowCut	Lo-Xovr	Hi-Xovr	SoftClip

3	LEVELS	low	mid	high
	Bnd-Lev			
	0dB ref			
	Meters			

4	COMPRES	low	mid	high
	Threshold			
	Ratio			
	Gain			
	Attack			
	Release			
	FeedFwd			
	Crest			

5	LIMITER	low	mid	high
	Threshold			
	Ratio			
	Attack			
	Release			
	FeedFwd			

6	EXPAND	low	mid	high
	Threshold			
	Ratio			
	Attack			
	Release			
	Range			

7	Par-Lnk	Norm-Dly



APPENDIX C

If problems or questions arise regarding your M5000, please check the following before you contact your dealer, TC Distributor or TC's head office in Denmark:

HARDWARE CONFIGURATION:				
FRAME SERIAL NO:				
	NO: 1	NO: 2	NO: 3	NO: 4
ADA-1 SERIAL NO:				
DSP-2 SERIAL NO:				
DSP-1 SERIAL NO:				
5DISK SERIAL NO:		DISK TYPE: 720 Kb/1.44 Mb		DD/HD
MEMORY CARD TYPE:		SIZE:		

SOFTWARE CONFIGURATION	
BIOS VERSION: *	
APPL. SOFTWARE: **	

M5000 CONNECTIONS				
Analog IN:	YES	NO	Balanced	Unbalanced
Digital IN:	YES	NO	AES/EBU	SPDIF RCA/OPT.
Analog OUT:	YES	NO	Balanced	Unbalanced
Digital OUT:	YES	NO	AES/EBU	SPDIF RCA/OPT.
WHAT KIND OF EQUIPMENT IS CONNECTED TO THE M5000 ?				

DESCRIBE THE PROBLEM AND IN WHICH SITUATION IT OCCURS:

* Refer to the 'SOFTWARE INSTALLATION'-module in the 'CONFIGURATION'-SECTION.

** Switch the M5000 OFF. During next 'power on', software version is shown in the display for a few seconds.

SELF TEST PROCEDURE IN BIOS 1.07 or higher M5000

The BIOS 1.07 (or higher) has built-in diagnostic test features. Hopefully you will never need them but they are implemented in order that the user can check the machine before it is sent for repair. Each time the M5000 is powered on, a quick test is done. These tests consist of the following steps:

All 4 LEDs on the CPU board are turned on.

Check BIOS EPROM checksum, if the checksum is bad LD1 on the CPU-board will turn on, and if the front panel is working then the preset LED's will show 'E01'. The M5000 will then halt.

Part of the dynamic RAM is tested, if the RAM is bad LD2 on the CPU-board will turn on, and if the front panel is working then the preset LED's will show 'E02'. The M5000 will then halt.

Contact to the LCD display is tested, if no contact is established LD1 and LD2 on the CPU-board will turn on, and if the front panel is working then the preset LEDs will show 'E03'. The M5000 will then halt.

LD4 on the CPU-board will stay on showing power is on.

If any problems occurs during operation of the M5000, e.g. disk problems or MIDI communication the user can select 2 different test sessions to be run.

SESSION 1: Total CPU test.

This session will run the following tests:

- 1. DYN RAM**
- 2. JEIDA MEMORY CARD SLOT**
- 3. EEPROM TEST**
- 4. EXTERNAL INTERRUPTS**
- 5. MIDI PORTS**
- 6. DISK DRIVE TEST**
- 7. MODULE CARD DETECTION (cannot detect DSP cards with a “+”)**

A MIDI cable must be connected from MIDI output to MIDI input in order to check the MIDI ports.

In order to test the disk drive, a formatted 720 Kb or 1.44 Mb disk must be inserted. **The data on the disk will be preserved.**

Keep BYPASS and EDIT pressed while turning power on. After a while the display will show:

M5000 diagnostics
Please wait

Now release the keys.

The test will run by it self and if any errors are detected user will be prompted and asked to take action. It will be shown on the LCD display.

Before the JEIDA test, the user will be prompted :

JEIDA test will destroy all data on card
Press DO to continue, UNDO to skip.

Insert a JEIDA memory card in the slot in order to check the Memory card slot.

If DO is pressed the data on the JEIDA card will be lost, if UNDO is pressed, this test will be skipped and the next test will be done.

When all tests are done and no errors were detected, the display will show :

Tests OK
Press DO to continue (*)

If any errors were detected the display will show:

Errors detected
Press DO to continue (*)

Pressing DO will result in the following message:

Push any key to detect Cards..
Then push any key to continue..

The M5000 will look for installed cards, and show the type and address of the detected cards.

If one ADDA and one DSP are installed the display will show:

ADDA at addr 1 DSP at addr 0

Pressing DO will enter the service card software mode. This is for future use. At this point all tests are done, and the M5000 should be powered down.

TROUBLESHOOTING/ERR. CODES**M5000**

ERROR	DESCRIPTION	ACTION
E01	EPROM checksum error (IC 31 & IC 32). The BIOS EPROMs may be defect or is badly connected in the socket.	Turn the M5000 off and on. If the error still is there, fill in the check form on page 1 and contact your dealer.
E02	Static RAM error (IC 22). The Static RAM may be defect or has a bad connection to the socket.	Turn the M5000 off and on. If the error still is there, fill in the check form on page 1 and contact your dealer.
E03	Bad contact between Display - CPU-board.	Turn the M5000 off and on. If the error still is there, fill in the check form on page 1 and contact your dealer.
E04	Internal error trap. A heavy line transient might cause these errors or bad internal connections.	Make note on the ALGO/PROGRAM you are running and the keys you pressed up to the error. Try to power off and the reestablish the error. If this is possible, please contact your dealer.
E05	Stack overflow in CPU (IC 4).	
E06	Multitask overflow in CPU (IC 4).	
#1 (LCD display)	EEPROM error (IC 14). Probably you will get a serial# type mismatch message as well on next power up.	Turn the M5000 off and on. If the error still is there, fill in the check form on page 1 and contact your dealer.
#2 (LCD display)	Flash PROM error (IC 23). The Flash PROM may be defect or has a bad connection in the socket.	Turn the M5000 off and on. If the error still is there, fill in the check form on page 1 and contact your dealer.
Serial information mismatch ...	RAM/Backup failure. Installed options are lost. Standard software can run with BIOS higher than 1.08.	Press as noted DO and write down the 16 character code and M5000 frame serial no (28 xx xx) and contact your dealer. In a tight situation you might press UNDO instead of DO and run the standard software (BIOS higher than 1.08), however, it may need to be re-installed. The error message will appear on every power up.
Device is hanging after ADA-1 was removed (1.13)	While the ADA-1 was present the I/O selector was still set to A/A&D. It will expect analog input and there isn't any.	Reinstall the ADA-1 module again change the I/O selector from A/A&D to D/D mode. Then you can remove the ADA-1 module. This problem was fixed in software version 1.14.

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APPENDIX D

TECHNICAL SPECIFICATIONS

M5000

All Specs is measured with ADA-1 STEREO ANALOG IN/OUT module installed.

Max. Input Level	@ - 8 dB gain, + 22,0 dBu @ 0 dB gain, + 14,8 dBu @ 12 dB gain, + 2,8 dBu
Input Impedance	20 KOhm, electronically balanced, pin 2+, 3-
Input Gain	± 12 dB
Input CMRR	DC - 1 KHz, > 60 dB 1 KHz - 20 KHz, > 40 dB
Max. Output Level	+ 22 dBu
Output Signal Balance	>40 dB @ 1 KHz (BBC method)
Output Impedance	100 Ohm, electronically balanced, floating type, pin 2+, 3-
Output Gain	-18 dB to + 12 dB
Frequency Response	10-22 KHz, +0 -1 dB, Fs=48.0KHz 10-20 KHz, +0 -1 dB, Fs=44.1KHz 10-15 KHz, +0 -0.5 dB, Fs=32.0KHz
Total Harmonic Dist.	< 0.03 %, 1 KHz, 0 dBu
Inter modulation Dist.	< 0.03 %
Dynamic Range	> 98 dB
Crosstalk	< -80 dB @ 1 KHz
Group Delay Linearity	< 5 μS
Phase Linearity	Better than 5°
Digital Conversion	Input: Delta Sigma 64x oversampling, 18 bit res. Output: Linear 8x oversampling, 20 bit res.
Sampling Rate	48.0 KHz, 44.1 KHz, 32.0 KHz
Environment	Operating 0° to 50°, storage -20° to 60°
Power Requirements	100 - 240 Vac, 50-60 Hz

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Power Consumption	20 - 60 watts, depending on configuration
Finish	Black anodized aluminum face plate. Black painted steel top and bottom plate. Chromatic steel chassis.
Net Weight	8.6 kg (19 lbs)
Shipping Weight	10 kg (22 lbs)

Due to continuous development, TC Electronic reserves the right to change specifications without further notice.

Power Switch	Rocker type
Device Selector	Selects the DSP to be controlled
Edit Page	Selects next or previous Edit Page
Dials A, B, C, D	Four dials for parameter editing
Program Dial	Control Program- recall, view and store
Do, Undo	Executes and cancels changes made
Program	Selects Program Algorithm Mode
Edit	Selects Edit Parameter Mode
Utility	Selects utility display
Bypass	Bypass of active devices
Load LED	Lit when parameters are updating
Timecode LED	Lit when receiving timecode
MIDI In LED	Lit when receiving MIDI
Digital In LED	Lit when receiving at digital inputs
LAN/SCSI LED	Lit when reading or writing data
Parameter Display	80 character alphanumeric display
Algorithm/Program	Displays algorithm type and program name
Program Number	3 digit program number display
Input Level Meter	Dual 10 segment LED

MIDI	In, Out and Thru
Remote	7-way custom RS-232 or RS485 In and Out, + power
Pedal	Programmable switch type, not implemented
SMPTE	Input for cue list management. The SMPTE jack plug must be an unbalanced connection with the TIP = HOT and the RING = GROUND . The SMPTE input accepts signals from -10 dBu and up
OPTION	For future options such as PCMCIA or SCSI, a Local Area Network option 2.5 Mbit/Sec. high speed data exchange between M5000 and Macintosh, Optical drive, Hard drive or another M5000

AES/EBU In/Out

XLR Professional Format. Sample rates between 32.0 KHz and 48.0 KHz

Optical In/Out

Optical Consumer Digital Format. Sample rates between 32.0 KHz and 48.0 KHz

SPDIF In/Out

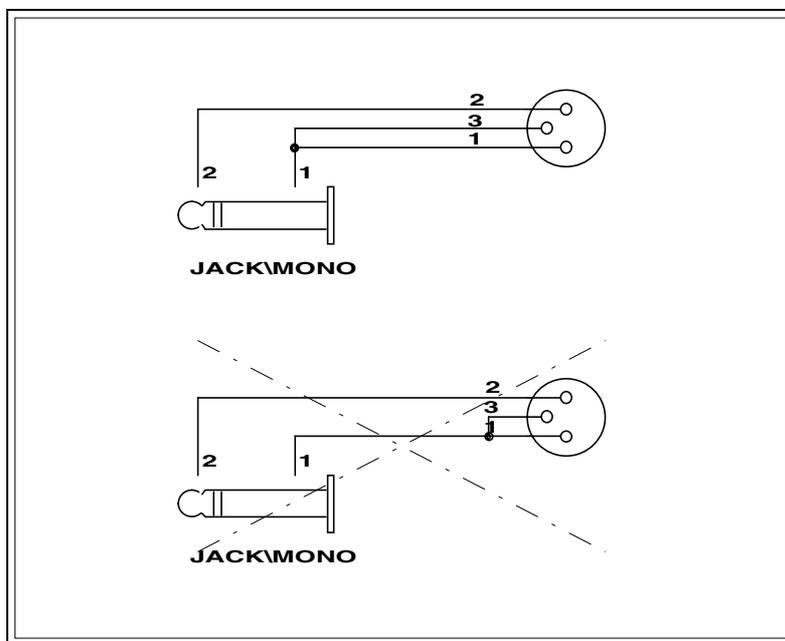
RCA Phono Consumer Digital Format. Sample rates between 32.0 KHz and 48.0 KHz

Left & Right Input

XLR 20 KOhm balanced. Max. input +22 dBu, pin 2 +, Pin 3 -.

Left & Right Output

XLR 100 Ohm balanced, floating type. Max. output +22 dBu, pin 2 +, Pin 3 -



To unbalance an input or output to the ADA-1 module, make that the the cable with the unbalancing pin 3/1 connection is made at the mono plug end of the cable, as shown on the figure. Pin 1 is the shield.

APPENDIX E

CABLES FOR DIGITAL AUDIO

M5000

In order to get a clean and noiseless digital signal flow the cable in which the digital signal is running has a great influence - especially over longer distances. Here is a list of cables recommended for digital interfaces by the corresponding manufacturer.

AES/EBU PROFESSIONAL DIGITAL AUDIO	
Manufacturer	Type
GOTHAM AG R'DORF, SWITZERLAND	GAC-2 (AES/EBU), 115ohm, +/-20%
NEGLEX – MOGAMI	3080 (AES/EBU), 110 ohm
GEPCO INT'L INC, CHICAGO	PN5524 – E131675 (ul), CM 24 AWG SHIELDED 75c
CANARE	105 AES/EBU
BELDEN	9860 (br. Sh.) 9271 (foil. Sh.), 124 ohm (Coaxial)
SPDIF CONSUMER DIGITAL AUDIO	
Manufacturer	Type
BELDEN	8217 OR 9259, 75OHM (Coaxial, RG-59/U-type)
TOSHIBA	TOCP174Y (OPTICAL)
SONY	POC-15 (OPTICAL)

Use always high-quality, low capacitance cables with fixed impedance (Coaxial), 110 Ω for AES/EBU and 75 Ω for SPDIF. There is no guarantee that it will work properly if an ordinary microphone cable is used for AES/EBU-communication or ordinary RCA cables for typical HI-FI equipment.

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APPENDIX F

TC BBS

M5000

The purpose of the TC Bulletin Board is to distribute new software, programs and presets for TC products and to share information between TC Electronic and the users of TC equipment.

In order to use the TC BBS you need the following:

1. An IBMtm, Ataritm or MACtm computer.
2. A communication program such as Procomm, Crosstalk or one of several public domain programs.
3. A modem, (a modem is an interface for your computer that enables you to connect your computer to another computer through the telephone line).

In the communication program you have to set certain parameters: i.e. (for the bulletin board in Denmark) 300-14400 Baud, 8 Data bits, No parity and 1 Stop bit. You get the best result if you set your program to use the ANSI terminal emulator.

Depending on where you are in the world, you can call the following numbers:

Bulletin Board	Number to call	Baud rate	Data bits	Parity	Stop bits
TC Denmark	+45 86 26 28 99	300-14400	8	N	1
Germany ¹	+49 40 45 80 90	300-19200	8	N	1
TC USA	805-374 9343	300-14400	8	N	1

Important!

Once you are connected to the bulletin board, you will be asked what the serial number of your M5000 frame is - so you better have that ready before calling, in order not to waste expensive on-line time while looking for the serial number on the M5000 - notice that you need the serial number from the frame - not the number from the modules - it will begin with 28x xxx.

When you are on-line, you will be guided through the menus and messages on the screen will explain what to do, when you want to download (receive) a program, read a message or leave a message etc.

On the bulletin board you will find the latest software version for the M5000 together with different utilities such as programs for dumping software to the M5000 from a computer through MIDI, program-files, newest information from TC and much more.

Call the bulletin board NOW and see for yourself...

¹ProAudio Net - A commercial BBS.

APPENDIX M

BIOS AND FLASH MINIMUM REQUIREMENT

The following table shows a connection between the released software versions and the BIOS versions. Also the required Flash EPROM size is shown:

SOFTWARE version	BIOS version	FLASH size
1.04	1.04	1 Megabit
1.06	1.04	1 Megabit
1.07	1.04	1 Megabit
1.09	1.04	1 Megabit
1.11	1.04	1 Megabit
1.12	1.04	1 Megabit
1.13	1.04	1 Megabit
1.14A (ATAC)	1.04	1 Megabit
1.15	1.04	1 Megabit
next release	2.00	2 Megabit

Note: Some of the features in software version 1.14 and higher will not be supported in BIOS version 1.04. A list of such features are found in the following table:

SOFTWARE version	FEATURES	Min. BIOS version	Min. FLASH size
x.xx	Self test proc.	1.07	1 Megabit
1.12	SAMPLING option	1.04	1 Megabit
1.13	MD2	1.04	1 Megabit
1.14	MD2	1.04	1 Megabit
1.14ATAC	ATAC support,	1.04*	1 Megabit
1.15	SAMPLER (SIMM) TOOLBOX** SMPTE PARAMETRIC EQ	2.00	2 Megabit

* Updating to BIOS 2.00 will improve ATAC performance

** Only if MD2 is installed

The software version is shown in the display during the power-on sequence of the M5000. The BIOS version and the Flash EPROM size are shown in the M5000 Setup Utility Menu. Refer to page 2 in the SOFTWARE INSTALLATION chapter in the CONFIGURATION section.

All M5000s with a higher serial number than 281 000 are all updated with BIOS version higher than 2.0 and 2 Megabit FLASH EPROM size !

The M5000 has standard MIDI In/Out/Thru connectors located on the rear panel. This chapter describes MIDI operation of the M5000, which you will see is quite extensive.

MIDI operation of the M5000 allows you to do the following:

- Control algorithm-parameters using MIDI Controllers.
- Recall programs using MIDI Program Change.
- Re-map programs (useful e.g. for associating a program with a synthesizer preset).
- Communicate using MIDI System-Exclusives (for software-developers... see the end of this chapter).

If you wish to fully understand MIDI as such, there are a number of books on the subject available at music stores. However, you will not need a thorough understanding of MIDI to utilize the features discussed in this chapter.

APPLICATIONS AND MIDI

As you know, multiple applications (DSP-cards) can be running at the same time. All applications share the same MIDI input/output, but it is vital to understand that each application has its own completely individual MIDI-setup. Each application has individual input/output channels and Program Maps. MIDI data received at MIDI In is actually fed into all applications, and MIDI output from all applications is merged and transmitted at MIDI Out.

Please note, that the System-Exclusive Device# (which is used to identify the M5000 when it is being controlled from a Macintosh or PC-editor) is set for the entire M5000 frame. You can still access each individual DSP-card, but this is controlled from the editor.

SETTING UP THE M5000 FOR MIDI OPERATION

Press the UTILITY button, and then turn the PROGRAM knob until the MIDI menu appears. Press the <Page button as many times as possible. You will now see the following (actual values may be different, but that is quite OK):

MIDI INPUT Page:

INPUT	CTL.IN	PRG.IN	PRG.BANK	MENU
ch1	on	on	ROM	MIDI



The **INPUT** knob selects on which channel the M5000 is to receive MIDI data. If the knob is turned fully counter-clockwise, 'omni-mode' is selected. In this case, the M5000 receives MIDI data on all channels.



When **CTL.IN** (Controller Input) is enabled, the M5000 will respond to MIDI Controller messages. Controllers are used for changing algorithm-parameters (more on this below in 'MIDI Controllers').



When **PRG.IN** (Program Input) is enabled, the M5000 will respond to MIDI Program Change commands. If the **PRG.MAP** (Program Map as described below) is enabled, any received Program Change is modified to select a different program as specified in the Program Map.



PRG.BANK selects which bank presets are recalled from with the Program Change command.

MIDI OUTPUT Page:

OUTPUT PRG.OUT OFFST-O OFFST-I MENU

ch 1 off -1 1 MIDI



The **OUTPUT** knob selects on which MIDI channel a MIDI Program-Change will be output, if a preset is recalled on the front panel.



The **PRG.OUT** knob selects whether or not a MIDI Program Change will be issued when a program is recalled.



OFFST-O (Output-offset)



OFFST-I (Input-offset).

OFFST-O and **OFFST-I** (output and input offsets) are provided, because different manufacturers implement program changes differently. It is a typical problem that your sequencer may number programs from 1 to 128, while your effect-processor may number programs from 0 to 127. The fix for this is the **offset**. The input-offset is added to the number of the program you wish to recall. With the settings shown above, program #5 is recalled when you ask for program #5 on your sequencer, which is the most natural way. Without the offset, program #4 would be recalled when you ask for program #5.

The output-offset works in the opposite way, because the program-change is going in the opposite direction.

You can also use the offsets to access a completely different range of programs. With an input-offset of 101, you can recall presets 101 and upwards by asking for presets 1 and upwards (which MIDI normally would make impossible).

PROGRAM CHANGES

When Program Input is enabled (see above), the M5000 will respond to Program Changes received via MIDI.

If the Program Map is disabled, the M5000 will respond normally to Program Changes (with regard to the offset, of course). Otherwise, the Program Map must be defined on the following page:

Program Map Page:

PRG.IN	MAPS TO PRESET	PRG.MAP	MENU
--------	----------------	---------	------

1	no chg.	off	MIDI
---	---------	-----	------



The **PRG.IN** knob decides which preset you would like to remap.



The **MAPS TO PRESET** selects which preset will be recalled when the **PRG.IN** preset is recalled. If 'no chg.' is selected, nothing will happen when the **PRG.IN** preset is recalled.



PRG.MAP enables/disables the program map.

The Program Map can be cleared on the following page:

MIDI Utility-Page:

SELECT FUNCTION	PRESS DO	MENU
-----------------	----------	------

CLEAR PRG. MAP	MIDI
----------------	------



Turn this knob to select one of the following functions:

CLEAR PRG.MAP

LOAD SETUP FROM CARD

SAVE SETUP TO CARD

LOAD SETUP FROM DISK

SAVE SETUP TO DISK

Press DO to engage the function.

MIDI CONTROLLERS

The M5000 allows you to control any parameter of any algorithm with a fixed set of controllers.

A MIDI controller is essentially a knob (such as a modulation wheel) which goes smoothly from 0 (min.) to 127 (max.). Real-time MIDI control allows you to control a given parameter (e.g. REVERB MIX, OUTPUT LEVEL, REVERB DECAY etc.) with a controller. In the following, each parameter of each algorithm is listed with the associated controller#.

Since MIDI controllers always go from 0 to 127, they are scaled to fit with the associated parameter. 0 represents the lowest possible value of the parameter, while 127 represents the highest possible value of the parameter. Setting REVERB MIX to 0 thus results in 0%, while 127 results in 100%.

REVERB-1 & REVERB-2:

Parameter	Controller#	Parameter	Controller#
MIX	10	HICUT	21
INLEV	11	ATT	22
OUTLEV	12	LO-XOVR	23
DECAY	13	HI-XOVR	24
x LOW	14	INITLEV	25
x HIGH	15	REVLEV	26
DIFFUSE	16	I-XFEED	27
SHAPE	17	REVDIFF •	28
x SIZE	18	BUILDUP •	29
PREDLY	19	IATTACK •	30
REVFEED	20	IDECAY •	31

Parameters marked with • are only available in REVERB-2.

REVERB-3:

Parameter	Controller#	Parameter	Controller#
MIX	10	LM-XOVR	19
INLEV	11	HI-XOVR	20
OUTLEV	12	PREDLY	21
DECAY	13	DISTANS	22
x LOW	14	HICUT	23
x LOMID	15	ATT	24
x HIGH	16	MODRATE	25
DIFFUSE	17	MODDPH	26
LO-XOVR	18	DIFTYPE	27

NONLIN-1:

Parameter	Controller#	Parameter	Controller#
MIX	10	LOCUT	17
INLEV	11	HICUT	18
OUTLEV	12	DIFFUSE	19
PREDLY	13	PREDIFF	20
ATTACK	14	DIFTYPE	21
HOLD	15	WIDTH	22
RELEASE	16		

CHORUS-1:

Parameter	Controller#	Parameter	Controller#
MIX	10	SPEED	16
INLEV	11	DEPTH	17
OUTLEV	12	FBLOCUT	18
PHASE	13	FBHICUT	19
DELAY	14	HICUT	20
FB	15	ATT	21

DELAY-1:

Parameter	Controller#	Parameter	Controller#
MIX	10	FB	15
INLEV	11	FBLOCUT	16
OUTLEV	12	FBHICUT	17
LDELAY	13	HICUT	18
RDELAY	14	ATT	19

DELAY-2:

Parameter	Controller#	Parameter	Controller#
MIX	10	DEPTH	22
INLEV	11	PHASE	23
OUTLEV	12	INV-PAN	24
DELAY1	13	FB1	25
DELAY2	14	FB2	26
HICUT	15	XFB12	27
ATT	16	XFB21	28
LEVEL1	17	LOFB	29
PAN1	18	HIFB	30
LEVEL2	19	LOXOVR	31
PAN2	20	HIXOVR	32
SPEED	21		

REVPITCH:

Parameter	Controller#	Parameter	Controller#
MIX	10	HICUT2	23
INLEV	11	ATT2	24
OUTLEV	12	FB1	25
PITCH1	13	FB2	26
FINE1	14	XFB12	27
PITCH2	15	XFB21	28
FINE2	16	AMBMIX	29
LEVEL1	17	PREDLY	30
PAN1	18	SHAPE	31
LEVEL2	19	SIZE	32
PAN2	20	PITCDLY	33
HICUT1	21	PITCCFT	34
ATT1	22		

PITCH-1:

Parameter	Controller#	Parameter	Controller#
MIX	10	HICUT2	23
INLEV	11	ATT2	24
OUTLEV	12	FB1	25
PITCH1	13	FB2	26
FINE1	14	XFB12	27
PITCH2	15	XFB21	28
FINE2	16	DELAY1	29
LEVEL1	17	DELAY2	30
PAN1	18	DGSPEED	31
LEVEL2	19	POLYSPD	32
PAN2	20	POLYDLY	33
HICUT1	21	DGFILT	34
ATT1	22		

PITCH-2:

Parameter	Controller#	Parameter	Controller#
MIX	10	HICUT	17
INLEV	11	ATT	18
OUTLEV	12	DGSPEED	19
PITCH	13	POLYSPD	20
FINE	14	POLYDLY	21
FB	15	DGFILT	22
DELAY	16		

AMBIENCE:

Parameter	Controller#	Parameter	Controller#
MIX	10	LOCUT	17
INLEV	11	LOATT	18
OUTLEV	12	HICUT	19
SHAPE	13	HIATT	20
SIZE	14	SPEED	21
PREDLY	15	DEPTH	22
WIDTH	16	PDLYMUL	23

TAPFAC-1:

Parameter	Controller#	Parameter	Controller#
MIX	10	LEVEL9	44
INLEV	11	LEVEL10	45
OUTLEV	12	LEVEL11	46
SCALE	13	LEVEL12	47
PREDLY	14	LEVEL13	48
WIDTH	15	LEVEL14	49
LASTTAP	16	LEVEL15	50
CURTAP	17	LEVEL16	51
DELAY1	18	LEVEL17	52
DELAY2	19	LEVEL18	53
DELAY3	20	PAN1	54
DELAY4	21	PAN2	55
DELAY5	22	PAN3	56
DELAY6	23	PAN4	57
DELAY7	24	PAN5	58
DELAY8	25	PAN6	59
DELAY9	26	PAN7	60
DELAY10	27	PAN8	61
DELAY11	28	PAN9	62
DELAY12	29	PAN10	63
DELAY13	30	PAN11	64
DELAY14	31	PAN12	65
DELAY15	32	PAN13	66
DELAY16	33	PAN14	67
DELAY17	34	PAN15	68
DELAY18	35	PAN16	69
LEVEL1	36	PAN17	70
LEVEL2	37	PAN18	71
LEVEL3	38	LOCUT	72
LEVEL4	39	LOATT	74
LEVEL5	40	HICUT	75
LEVEL6	41	HIATT	76
LEVEL7	42	SPEED	77
LEVEL8	43	DEPTH	78

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t.c. electronic

DYNAMIC1:

Parameter	Controller#	Parameter	Controller#
MIX	10	M-LIMREL	45
INLEV	11	M-EXPTHR	46
OUTLEV	12	M-EXPRATIO	47
BALANCE	13	M-EXPATCK	48
LOWCUT	14	M-EXPREL	49
LMXOVR	15	M-EXPRANGE	50
MHXOVR	16	M-LEVEL	51
SOFTCLIP	17	M-CREST	52
L-COMTHR	18	M-DELAY	53
L-COMRATIO	19	M-LIMDLY	54
L-COMATCK	20	M-SFTKNEE	55
L-COMREL	21	M-METERS	56
L-LIMTHR	22	M-REF0DB	57
L-LIMRATIO	23	H-COMTHR	58
L-LIMATCK	24	H-COMRATIO	59
L-LIMREL	25	H-COMATCK	60
L-EXPTHR	26	H-COMREL	61
L-EXPRATIO	27	H-LIMTHR	62
L-EXPATCK	28	H-LIMRATIO	63
L-EXPREL	29	H-LIMATCK	64
L-EXPRANGE	30	H-LIMREL	65
L-LEVEL	31	H-EXPTHR	66
L-CREST	32	H-EXPRATIO	67
L-DELAY	33	H-EXPATCK	68
L-LIMDLY	34	H-EXPREL	69
L-SFTKNEE	35	H-EXPRANGE	70
L-METERS	36	H-LEVEL	71
L-REF0DB	37	H-CREST	72
M-COMTHR	38	H-DELAY	73
M-COMRATIO	39	H-LIMDLY	74
M-COMATCK	40	H-SFTKNEE	75
M-COMREL	41	H-METERS	76
M-LIMTHR	42	H-REF0DB	77
M-LIMRATIO	43	PARLNK	78
M-LIMATCK	44	NOMDELAY	79

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TOOLBOX:

Parameter	Controller#	Parameter	Controller#
MIX	10	EQTYPE2	40
INLEV	11	EQFREQ2	41
OUTLEV	12	PWIDTH2	42
INSON	13	NWIDTH2	43
BAL	14	PGAIN2	44
MONO	15	NGAIN2	45
LRSWAP	16	EQON2	46
ID_PHASE	17	EQTYPE3	47
DITHER	18	EQFREQ3	48
DITYP	19	PWIDTH3	49
MSON	20	NWIDTH3	50
MSANGLE	21	PGAIN3	51
FADECURVE	22	NGAIN3	52
FADER	23	EQON3	53
METER	24	EQTYPE4	54
RANGE	25	EQFREQ4	55
TICKS	26	PWIDTH4	56
HOLD	27	NWIDTH4	57
LDELAY	28	SSLOPE4	58
RDELAY	29	CSLOPE4	59
EQTYPE1	30	PGAIN4	60
EQFREQ1	31	NGAIN4	61
PWIDTH1	32	SGAIN4	62
NWIDTH1	33	EQON4	63
SSLOPE1	34	LPPM	64
CSLOPE1	35	RPPM	65
PGAIN1	36	CORR	66
NGAIN1	37	CORLEG	67
SGAIN1	38	FADVAL	68
EQON1	39		

SYSTEM-EXCLUSIVES

System-Exclusives (Sysex for short) is a subset of the MIDI-protocol, which allows software-developers (who are writing a Mac or PC-based editor) to communicate with the M5000 in a very technical manner (giving total control over the M5000). The Sysex-documentation is of a very technical nature, which is why it isn't described in this manual. The M5000 System-Exclusive Manual is available at any TC-office.

MIDI System-Exclusive Page:

SYSEX ID#	MENU
0	MIDI



Turn the **Device#** knob to set the device-number of this M5000.

The **Device#** is all you'll ever need to know about Sysex. This number must be set to same value both on your Macintosh/PC-editor and on your M5000 in order for them to be able to 'find' each other.

GUIDED TOURS

The following are step by step methods of familiarizing yourself with the various features of the M5000. They are intended to provide the user with sufficient working knowledge of all aspects of the unit in a condensed form. For a more in-depth explanation of the various parameters, please consult the appropriate chapters in the manual referred to in brackets (SECTIONNAME, MODULENAME).

- ✓ #1: SOFTWARE UPDATE
- ✓ #2: PROGRAM HANDLING
- ✓ #3: DISK/CARD HANDLING

How do I update with a new software ? (CONFIGURATION, SOFTWARE INSTALLATION)

1. Make sure that the M5000 is switched off.
2. Switch on power while holding the **BYPASS** button until the following appears:

M5000 SETUP UTILITY

choose option and press Do : Load (DISK)

3. Insert floppy disk into the disc drive or memory card in the memory card slot with new software.
4. Press **DO** until "**Select file to load :**" appears.
5. Turn the PROGRAM knob to select the software you want to load and press **DO** twice.
6. Wait while FLASH EPROM is updating.
7. Power-down and then power-up to initialize the new software as instructed in the display.

How do I call up a program: (GENERAL INSTRUCTION, PROGRAM HANDLING)

1. Press the **PROGRAM** button.
2. Turn knob **A** and choose the Memory Bank **SOURCE** where the program is stored, e.g. **ROM** , **RAM** or **FILE**.
3. Turn the **PROGRAM** knob to choose a program. (The **PROGRAM NUMBER** will blink).
4. Press **DO**. The program is now loaded and the **PROGRAM NUMBER** stops blinking.

How do I edit and store a program ? (GENERAL INSTRUCTION, PROGRAM HANDLING)

1. Press the **EDIT** button whilst in the program you want to edit.
2. By turning the knobs **A** to **D** and pressing the **PAGE** buttons left or right, all parameters can be accessed and changed as required. A red LED will appear in the **PROGRAM NUMBER** display next to the word "**EDITED**" to inform you that parameters in this program have been changed.
3. Press the **PROGRAM** button.
4. Turn knob **A** until >>**Ram**<< appears.
5. Turn knob **D** until >>**Store**<< appears and choose a new user preset number (RAM) for the program by turning the **PROGRAM** knob.
6. Press **DO** to store the new program in RAM.

How do I format a Floppy Disk/Memory Card ?**During a session:** (GENERAL INSTRUCTION, UTILITY HANDLING)

1. Insert floppy disk in the disc drive or memory card in the memory card slot.
2. Press the **UTILITY** button and turn the **PROGRAM** knob until the MENU >>**FILE**<< appears.
3. Choose which medium you wish to format by turning knob **A** until >>**FORMAT DISK**<< (floppy disc) or >>**FORMAT CARD**<< (PCMCIA-card) appears and press **DO**.
4. Turn knob **A** to select 720 Kb or 1.44 Mb formatting for disk size or 64 - 1024 Kb formatting for card size.
5. Press **DO** twice.

How do I save my programs to disk/card ? (GENERAL INSTRUCTION, PROGRAM HANDLING)

RAM programs:

1. Press the **PROGRAM** button.
2. Press the right **PAGE** button twice.
3. Turn knob **A** until >>**Ram to File**<< appears and then press **DO**.
4. Press **DO** when the file menu reads >>**Save Disk**<<.
5. Now give a name to the 'bank' of programs, which are in the FILE buffer.
6. Press **DO** to store the FILE buffer to disk.

FILE programs: Repeat 1-6, except 3.

How do I load programs into the M5000 ? (GENERAL INSTRUCTION, PROGRAM HANDLING)**Loading From a floppy disk:**

1. Press the **PROGRAM** button.
2. Press the right **PAGE** button twice.
3. Turn knob **A** until >>**Load Disk**<< appears and then press **DO**.
4. Select the file to load by turning the **PROGRAM** knob and then press **DO**.

Loading From a Memory Card:

1. Press the **PROGRAM** button.
2. Press the right **PAGE** button twice.
3. Turn knob **A** until "**Load Card**" appears and then press **DO**.
4. Select the file to load by turning the **PROGRAM** knob and pressing **DO**.



MIDI System-Exclusive Documentation

Revision 2.00

(16/06/97)

Documents all features in application-software APL115.M5K

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1 Introduction

1.1 Overview

The M5000 sends and receives system-exclusive messages (sysex). The sysex-protocol gives you access to the following:

- Each individual parameter in each algorithm
- System-parameters (audio-routing, formats, sample-rates, meters etc.)
- System-information (software version, installed options)
- Preset-handling (preset-transfers)
- Preset-selection (to recall presets without needing to know about MIDI-channels)

1.1.1 The Parameter-Access Dump

The Parameters-Access dump is used for almost all communication to and from the M5000, and is therefore the most extensive part of the protocol. Parameters that are specific to individual algorithms are described in chapter 5. Parameters that are general to the system (DSP-cards) are described in chapter 5.

There are many different parameter-types in the M5000, including milliseconds, hertz, decibels, tables and character-strings. It would be nearly impossible to describe and list each parameter-type in detail, which is why TC supplies a C programming-interface to assist you in displaying the correct value for each individual parameter-type (eg. "10 kHz" or "50%"). The programming-interface consists of two files, "CLASS.C" and "CLASS.H". The two files can be downloaded from the TC User-Club BBS on the following phone-numbers:

Denmark: +45 - 86 21 75 99
 USA: +1 - 805 373 1828

Please refer to chapter 0 for information on how to use the interface.

1.1.2 System-Configuration/Info Dumps

Information about the system (software version/installed options/DSP-cards etc.) can be requested from the M5000. This is described in chapter 0.

1.1.3 Preset-Handling Dumps

These dumps provide you the means to store, recall, dump and retrieve presets as well as request information about them. This is described in chapter 0.

1.2 General Format

Sysex-packets are transferred to and from the M5000 using the following general format:

```

Sysex-start    $f0
TC ID          $33
Device#        $00-$7f
Card#          $01-$04
Packet-type    $00-$07

Data specific to the Packet-type

Sysex-end      $f7
  
```

The Device# must correspond with the Device# set for the M5000 frame.

The Card# refers to the ID of each consecutive DSP-card or layer. A value of 1 refers to the first DSP-card and a value of 2 refers to the second DSP-card. A value of 0 is only used in certain operations that refer to the entire M5000 frame.

The Packet-type signifies the type of packet. Each individual request and dump has its own unique packet-type. The following packet-types are transmitted and/or recognized by the M5000:

\$00	Set parameter(s)
\$01	Request parameter(s)
\$02	Recall Preset
\$03	Request Preset Info
\$04	Request System Configuration Info
\$05	Preset Info

The M5000 is very tolerant about incomplete or erroneous sysex-packets, but it is still recommended that you keep your packets clean with all values within range. The M5000 allows packet-sizes of any size (which is relevant for requesting a large number of parameters), though you need to obey the precautions regarding the parameter-queue as described in the next chapter.

2 Individual Parameter-Access

2.1 Overview

Each individual parameter in the M5000 has a unique ID. This gives you access to each parameter in each algorithm and general system-parameters, such as Input-gain or Bypass, for instance.

Not all parameters exist at the same time. For example, parameters in the REVERB-3 algorithm don't exist if a PITCH-1 algorithm is running on the DSP-card in question. Trying to set nonexistent parameters will have no effect, and requesting their setting will produce no result.



Note: It is possible to obtain information about the algorithm currently running, to determine which parameters should be polled. Please refer to Chapter 0 for a description of how to obtain information about the algorithm currently running on a DSP-card.

2.2 Parameter-IDs and Values

All parameter-IDs and values in the M5000 are 14 bit wide. In addition, parameter-values are signed, to allow for negative values. The two ranges are as follows:

Parameter-IDs: 0 to 16383 (\$0000 to \$3fff)
 Parameter-values: -8192 to 8191 (-\$2000 to \$1fff)

2.2.1 Parameter-IDs

In the following documentation, the 14-bit parameter-IDs are shown as <Par# xxxx>, although their physical placement in the sysex-packet is as follows:

```

    Par #xx__      bit 8-13 (MSB First)
    Par #__xx      bit 0-7  (LSB Last)
...is shown as:
    <Par #xxxx>
  
```

In sections 0 and 0 you'll find two C-routines that convert the two MIDI-bytes to a single C-type unsigned short and vice versa.

2.2.2 Parameter-values

In the following documentation, the 14-bit signed parameter-values are shown as <Value #xxxx>, although their physical placement in the sysex-packet is as follows:

```

    Value #xx__    bit 8-13 (MSB First, sign in bit 13)
    Value #__xx    bit 0-7  (LSB Last)
...is shown as:
    <Value #xxxx>
  
```

In order to convert these double MIDI-bytes to a single C-type short, the sign bit must be extended from bit 13 to bit 15. In sections 0 and 0 you'll find two C-routines that convert the two MIDI-bytes to a single C-type short and vice versa.

2.3 Requesting Parameter Values

The following sysex-packet allows you to request the setting of a number of parameters. In a single packet, you can request as many or as few parameters as you like.

```

Sysex-start    $f0
TC ID          $33
Device#        xx
Card#          xx
Packet-type    $01 - Request
<Par #xxxx>
<Par #yyyy>
<Par #zzzz>
...
...
Sysex-end      $f7

```

The M5000 replies with a Parameter Dump, as described next:

2.4 Setting Parameter Values

In a single parameter-dump, you can set as many or as few parameters as you like. In order to minimize MIDI-traffic, you should set as many parameters as possible in a single dump.

```

Sysex-start    $f0
TC ID          $33
Device#        xx
Card#          xx
Packet-type    $00 - Dump
<Par #xxxx>
<Par Value>
<Par #yyyy>
<Par Value>
<Par #zzzz>
<Par Value>
...
...
Sysex-end      $f7

```

2.4.1 Truncation and Mutual Dependencies

If a parameter-value is out of range, it is truncated to fit. Please note, that some parameters (such as cross-overs) have floating minimum and maximum values. This scheme follows a fairly simple logic, although you must implement this yourself in order to track the correct value for the user; the M5000 has no way of telling you that a parameter-value has been truncated. These mutual dependencies are described as necessary in conjunction with the parameter-listings in chapter 0 and 0.

2.4.2 Linked Parameters

Some parameters are linked to always contain the same value (a good example of this is the `0dBRef` parameter in the `DYNAMIC1` algorithm). Generally, you shouldn't display or manipulate more than one of the linked parameters. With the `0dBRef` example, simply choose one of the parameters as the only one.

2.4.3 The Parameter-Queue

The M5000 places all parameters that need to be changed in a queue. Some parameters take a little time to recalculate, while others change instantaneously. The parameters are extracted from the queue as fast as possible.



Note: If you set a parameter that hasn't yet been extracted from the queue, the queue-entry for the given parameter is updated to hold the new value. This means that you don't need to worry about placing delays in the MIDI data-stream while the user is dragging a slider in a patch-editor. Simply transmit the new value for the given parameter every time the slider is moved.



The parameter-queue in the M5000 holds 32 messages. If you are setting up an algorithm like TAPFAC-1 or DYNAMIC1 (which have more than 32 parameters), you must place slight delays in the MIDI data-stream. Some special parameters take some time to recalculate (you will know these parameters from the M5000 front panel). Instead of having a specific delay for each parameter-type, you should simply transmit the parameters at a pace that works.

2.5 Spontaneous Data-Emissions

The M5000 will generally never output parameter-packets spontaneously. However Recall Preset packet will be transmitted if the user recalls a preset via the front-panel or an ATAC.

Meters are never transmitted spontaneously, but must be polled.

2.6 Conversion Routines

2.6.1 convertMIDItoPar

The following routine combines the two MIDI-bytes that identify the parameter-ID and return a single unsigned short:

```
unsigned short convertMIDItoPar(char byte1, char byte2)
{
    return byte2+(byte1 << 7);
}
```

2.6.2 convertPartoMIDI

The following routine derives the two MIDI-bytes that identify the parameter-ID from a single short:

```
void convertMIDItoPar(unsigned short parNo, char *byte1,
                    char *byte2)
{
    *byte1=parNo >> 7;
    *byte2=parNo & 0x7f;
}
```

2.6.3 convertMIDItoValue

The following routine combines the two MIDI-bytes that identify the parameter-value and return a single short:

```
short convertMIDItoValue(char byte1, char byte2)
{
    short value;

    value=(byte2 & 0x7f) + ( (short) (byte1 & 0x7f) << 7);
    if (value & 0x2000) i |= 0xc000; // Extend sign bit
    return value;
}
```

2.6.4 convertValuetoMIDI

The following routine derives the two MIDI-bytes that identify the parameter-ID from a single short:

```
void convertValuetoMIDI(short value, char *byte1, char *byte2)
{
    *byte1=(value >> 7) & 0x7f;
    *byte2=value & 0x7f;
}
```

3 Algorithm-Parameters

3.1 REVERB-1 & REVERB-2

Parameter-name	ID (Hex)	Min (Dec)	Max (Dec)	Class (Type)
MIX	1000	0	100	CLS_PERCENT
INLEV	1001	0	100	CLS_DB1
OUTLEV	1002	0	100	CLS_DB1
DECAY	1003	3	600	CLS_SEC1
x LOW	1004	1	250	CLS_NON2
x HIGH	1005	1	200	CLS_NON2
DIFFUSE	1006	1	25	CLS_NON0
SHAPE	1007	0	5	CLS_SHAPE0
x SIZE	1008	0	20	CLS_SIZE0
PREDLY	1009	0	2000	CLS_MS1
REVFEEED	100A	0	1000	CLS_MS1
HICUT	100B	14	30	CLS_FRQ0
ATT	100C	20	100	CLS_DB1
LO-XOVR	100D	0	30	CLS_FRQ0
HI-XOVR	100E	0	30	CLS_FRQ0
INITLEV	100F	0	100	CLS_DB1
REVLEV	1010	0	100	CLS_DB1
RWIDTH	1011	0	100	CLS_PERCENT
I-XFEED	1012	0	1	CLS_OFFON

These last 4 parameters are only available in REVERB-2:



REVDIFF	1013	0	100	CLS_PERCENT
BUILDUP	1014	0	100	CLS_NON0
IATTACK	1015	0	100	CLS_DB1
IDECAY	1016	0	100	CLS_DB1

3.2 REVERB-3

Parameter-name	ID (Hex)	Min (Dec)	Max (Dec)	Class (Type)
MIX	1500	0	100	CLS_PERCENT
INLEV	1501	0	100	CLS_DB1
OUTLEV	1502	0	100	CLS_DB1
DECAY	1503	3	300	CLS_SEC1
x LOW	1504	1	250	CLS_NON2
x LOMID	1505	1	200	CLS_NON2
x HIGH	1506	1	200	CLS_NON2
DIFFUSE	1507	1	99	CLS_NON0
LO-XOVR	1508	0	23	CLS_FRQ0

LM-XOVR	1509	10	25	CLS_FRQ0
HI-XOVR	150A	20	30	CLS_FRQ0
PREDLY	150B	1	200	CLS_MS1
DISTANS	150C	0	15	CLS_NON0
HICUT	150D	14	30	CLS_FRQ0
ATT	150E	20	100	CLS_DB1
MODRATE	150F	1	200	CLS_NON0
MODDPH	1510	0	100	CLS_PERCENT
DIFTYPE	1511	0	4	CLS_DIFF0

3.3 NONLIN-1

Parameter-name	ID (Hex)	Min (Dec)	Max (Dec)	Class (Type)
MIX	1600	0	100	CLS_PERCENT
INLEV	1601	0	100	CLS_DB1
OUTLEV	1602	0	100	CLS_DB1
PREDLY	1603	0	500	CLS_MS0
ATTACK	1604	0	500	CLS_MS0
HOLD	1605	10	500	CLS_MS0
RELEASE	1606	0	500	CLS_MS0
LOCUT	1607	0	20	CLS_FRQ0
HICUT	1608	16	30	CLS_FRQ0
DIFFUSE	1609	0	25	CLS_NON0
PREDIFF	160A	0	100	CLS_NON0
DIFTYPE	160B	0	3	CLS_PREDIFF
WIDTH	160C	0	100	CLS_PERCENT

3.4 CHORUS-1

Parameter-name	ID (Hex)	Min (Dec)	Max (Dec)	Class (Type)
MIX	1100	0	100	CLS_PERCENT
INLEV	1101	0	100	CLS_DB1
OUTLEV	1102	0	100	CLS_DB1
PHASE	1103	0	2	CLS_PHASE1
DELAY	1104	1	670	CLS_MS0
FB	1105	0	99	CLS_PERCENT
SPEED	1106	0	40	CLS_SPEEDS0
DEPTH	1107	0	100	CLS_PERCENT
FBLOCUT	1108	0	4	CLS_LOCUTS
FBHICUT	1109	0	4	CLS_HICUTS
HICUT	110A	14	30	CLS_FRQ0
ATT	110B	20	100	CLS_DB1

3.5 DELAY-1

Parameter-name	ID (Hex)	Min (Dec)	Max (Dec)	Class (Type)
MIX	1200	0	100	CLS_PERCENT
INLEV	1201	0	100	CLS_DB1
OUTLEV	1202	0	100	CLS_DB1
LDELAY	1203	1	670	CLS_MS0
RDELAY	1204	1	670	CLS_MS0
FB	1205	0	99	CLS_PERCENT
FBLOCUT	1206	0	4	CLS_LOCUTS
FBHICUT	1207	0	4	CLS_HICUTS
HICUT	1208	14	30	CLS_FRQ0
ATT	1209	20	100	CLS_DB1

3.6 DELAY-2

Parameter-name	ID (Hex)	Min (Dec)	Max (Dec)	Class (Type)
MIX	1300	0	100	CLS_PERCENT
INLEV	1301	0	100	CLS_DB1
OUTLEV	1302	0	100	CLS_DB1
DELAY1	1303	1	670	CLS_MS0
DELAY2	1304	1	670	CLS_MS0
HICUT	1305	14	30	CLS_FRQ0
ATT	1306	20	100	CLS_DB1
LEVEL1	1307	0	100	CLS_DB1
PAN1	1308	0	100	CLS_PANL
LEVEL2	1309	0	100	CLS_DB1
PAN2	130A	0	100	CLS_PANR
SPEED	130B	0	40	CLS_SPEEDS0
DEPTH	130C	0	100	CLS_PERCENT
PHASE	130D	0	2	CLS_PHASE1
INV-PAN	130E	0	1	CLS_ONOFF
FB1	130F	-100	100	CLS_PERCENT
FB2	1310	-100	100	CLS_PERCENT
XFB12	1311	-100	100	CLS_PERCENT
XFB21	1312	-100	100	CLS_PERCENT
LOFB	1313	20	100	CLS_DB1
HIFB	1314	20	100	CLS_DB1
LOXOVR	1315	0	30	CLS_FRQ0
HIXOVR	1316	0	30	CLR_FRQ0

3.7 REVPITCH

Parameter-name	ID (Hex)	Min (Dec)	Max (Dec)	Class (Type)
MIX	1700	0	100	CLS_PERCENT
INLEV	1701	0	100	CLS_DB1
OUTLEV	1702	0	100	CLS_DB1
PITCH1	1703	-12	12	CLS_NON0
FINE1	1704	-50	50	CLS_NON0
PITCH2	1705	-12	12	CLS_NON0
FINE2	1706	-50	50	CLS_NON0
LEVEL1	1707	0	100	CLS_DB1
PAN1	1708	0	100	CLS_PANL
LEVEL2	1709	0	100	CLS_DB1
PAN2	170A	0	100	CLS_PANR
HICUT1	170B	14	30	CLS_FRQ0
ATT1	170C	20	100	CLS_DB1
HICUT2	170D	14	30	CLS_FRQ0
ATT2	170E	20	100	CLS_DB1
FB1	170F	0	100	CLS_PERCENT
FB2	1710	0	100	CLS_PERCENT
XFB12	1711	0	100	CLS_PERCENT
XFB21	1712	0	100	CLS_PERCENT
AMBMIX	1713	0	100	CLS_PERCENT
PREDLY	1714	0	1500	CLS_MS1
SHAPE	1715	0	6	CLS_SHAPE0
SIZE	1716	0	20	CLS_SIZE0
PITCDLY	1717	10	40	CLS_MS0
PITCCFT	1718	5	100	CLS_NON0

3.8 PITCH-1

Parameter-name	ID (Hex)	Min (Dec)	Max (Dec)	Class (Type)
MIX	1800	0	100	CLS_PERCENT
INLEV	1801	0	100	CLS_DB1
OUTLEV	1802	0	100	CLS_DB1
PITCH1	1803	-12	12	CLS_NON0
FINE1	1804	-1200	1200	CLS_NON0
PITCH2	1805	-12	12	CLS_NON0
FINE2	1806	-1200	1200	CLS_NON0
LEVEL1	1807	0	100	CLS_DB1
PAN1	1808	0	100	CLS_PANL
LEVEL2	1809	0	100	CLS_DB1
PAN2	180A	0	100	CLS_PANR

HICUT1	180B	14	30	CLS_FRQ0
ATT1	180C	20	100	CLS_DB1
HICUT2	180D	14	30	CLS_FRQ0
ATT2	180E	20	100	CLS_DB1
FB1	180F	0	100	CLS_PERCENT
FB2	1810	0	100	CLS_PERCENT
XFB12	1811	0	100	CLS_PERCENT
XFB21	1812	0	100	CLS_PERCENT
DELAY1	1813	0	310	CLS_MS0
DELAY2	1814	0	310	CLS_MS0
DGSPEED	1815	5	50	CLS_NON2
POLYSPD	1816	5	50	CLS_NON0
POLYDLY	1817	5	18	CLS_NON0
DGFILT	1818	0	3	CLS_DGFILTS

3.9 PITCH-2

Parameter-name	ID (Hex)	Min (Dec)	Max (Dec)	Class (Type)
MIX	1900	0	100	CLS_PERCENT
INLEV	1901	0	100	CLS_DB1
OUTLEV	1902	0	100	CLS_DB1
PITCH	1903	-12	12	CLS_NON0
FINE	1904	-1200	1200	CLS_NON0
FB	1905	0	100	CLS_PERCENT
DELAY	1906	0	310	CLS_MS0
HICUT	1907	14	30	CLS_FRQ0
ATT	1908	20	100	CLS_DB1
DGSPEED	1909	5	50	CLS_NON2
POLYSPD	190A	5	50	CLS_NON0
POLYDLY	190B	5	18	CLS_NON0
DGFILT	190C	0	3	CLS_DGFILTS

3.10 TAPFAC-1

Parameter-name	ID (Hex)	Min (Dec)	Max (Dec)	Class (Type)
MIX	1B00	0	100	CLS_PERCENT
INLEV	1B01	0	100	CLS_DB1
OUTLEV	1B02	0	100	CLS_DB1
SCALE	1B03	1	100	CLS_PERCENT
PREDLY	1B04	0	1000	CLS_MS0
WIDTH	1B05	0	100	CLS_PERCENT
LASTTAP	1B06	1	18	CLS_NON0
CURTAP	1B07	1	18	CLS_NON0

DELAY1	1B08	0	6230	CLS_MS0
DELAY2	1B09	0	6230	CLS_MS0
DELAY3	1B0A	0	6230	CLS_MS0
DELAY4	1B0B	0	6230	CLS_MS0
DELAY5	1B0C	0	6230	CLS_MS0
DELAY6	1B0D	0	6230	CLS_MS0
DELAY7	1B0E	0	6230	CLS_MS0
DELAY8	1B0F	0	6230	CLS_MS0
DELAY9	1B10	0	6230	CLS_MS0
DELAY10	1B11	0	6230	CLS_MS0
DELAY11	1B12	0	6230	CLS_MS0
DELAY12	1B13	0	6230	CLS_MS0
DELAY13	1B14	0	6230	CLS_MS0
DELAY14	1B15	0	6230	CLS_MS0
DELAY15	1B16	0	6230	CLS_MS0
DELAY16	1B17	0	6230	CLS_MS0
DELAY17	1B18	0	6230	CLS_MS0
DELAY18	1B19	0	6230	CLS_MS0
LEVEL1	1B1A	0	100	CLS_PERCENT
LEVEL2	1B1B	0	100	CLS_PERCENT
LEVEL3	1B1C	0	100	CLS_PERCENT
LEVEL4	1B1D	0	100	CLS_PERCENT
LEVEL5	1B1E	0	100	CLS_PERCENT
LEVEL6	1B1F	0	100	CLS_PERCENT
LEVEL7	1B20	0	100	CLS_PERCENT
LEVEL8	1B21	0	100	CLS_PERCENT
LEVEL9	1B22	0	100	CLS_PERCENT
LEVEL10	1B23	0	100	CLS_PERCENT
LEVEL11	1B24	0	100	CLS_PERCENT
LEVEL12	1B25	0	100	CLS_PERCENT
LEVEL13	1B26	0	100	CLS_PERCENT
LEVEL14	1B27	0	100	CLS_PERCENT
LEVEL15	1B28	0	100	CLS_PERCENT
LEVEL16	1B29	0	100	CLS_PERCENT
LEVEL17	1B2A	0	100	CLS_PERCENT
LEVEL18	1B2B	0	100	CLS_PERCENT
PAN1	1B2C	0	20	CLS_PANG
PAN2	1B2D	0	20	CLS_PANG
PAN3	1B2E	0	20	CLS_PANG
PAN4	1B2F	0	20	CLS_PANG
PAN5	1B30	0	20	CLS_PANG
PAN6	1B31	0	20	CLS_PANG
PAN7	1B32	0	20	CLS_PANG
PAN8	1B33	0	20	CLS_PANG
PAN9	1B34	0	20	CLS_PANG
PAN10	1B35	0	20	CLS_PANG
PAN11	1B36	0	20	CLS_PANG
PAN12	1B37	0	20	CLS_PANG
PAN13	1B38	0	20	CLS_PANG
PAN14	1B39	0	20	CLS_PANG
PAN15	1B3A	0	20	CLS_PANG
PAN16	1B3B	0	20	CLS_PANG
PAN17	1B3C	0	20	CLS_PANG
PAN18	1B3D	0	20	CLS_PANG

LOCUT	1B3E	0	17	CLS_FRQ0
LOATT	1B3F	20	100	CLS_DB1
HICUT	1B40	17	30	CLS_FRQ0
HIATT	1B41	20	100	CLS_DB1
SPEED	1B42	0	40	CLS_SPEEDS0
DEPTH	1B43	0	100	CLS_PERCENT

3.11 AMBIENCE

Parameter-name	ID (Hex)	Min (Dec)	Max (Dec)	Class (Type)
MIX	1C00	0	100	CLS_PERCENT
INLEV	1C01	0	100	CLS_DB1
OUTLEV	1C02	0	100	CLS_DB1
SHAPE	1C03	0	5	CLS_SHAPE0
SIZE	1C04	0	20	CLS_SIZE0
PREDLY	1C05	0	1000	CLS_MS1
WIDTH	1C06	0	100	CLS_PERCENT
LOCUT	1C07	0	17	CLS_FRQ0
LOATT	1C08	20	100	CLS_DB1
HICUT	1C09	17	30	CLS_FRQ0
HIATT	1C0A	20	100	CLS_DB1
SPEED	1C0B	0	40	CLS_SPEEDS0
DEPTH	1C0C	0	100	CLS_PERCENT
PDLYMUL	1C0D	0	1	CLS_DLYMUL

3.12 DYNAMIC1

Parameter-name	ID (Hex)	Min (Dec)	Max (Dec)	Class (Type)
MIX	1A00	0	100	CLS_PERCENT
INLEV	1A01	0	100	CLS_DB1
OUTLEV	1A02	0	100	CLS_DB1
BALANCE	1A03	0	100	CLS_PANL
LOWCUT	1A04	0	21	CLS_FRQ1
LMXOVR	1A05	0	29	CLS_FRQ2
MHXOVR	1A06	0	29	CLS_FRQ3
SOFTCLIP	1A07	0	1	CLS_ONOFF
Low Section → COMTHR	1A08	13	124	CLS_DB1
COMRATIO	1A09	0	15	CLS_RATIO1
COMATCK	1A0A	0	15	CLS_DYNATCK
COMREL	1A0B	0	15	CLS_DYNDEC
LIMTHR	1A0C	76	100	CLS_DB1
LIMRATIO	1A0D	0	1	CLS_RATIO3
LIMATCK	1A0E	0	15	CLS_LIMATCK
LIMREL	1A0F	0	15	CLS_DYNDEC

	EXPTHR	1A10	2	100	CLS_CLS_DB1
	EXPRATIO	1A11	0	15	CLS_RATIO2
	EXPATCK	1A12	0	15	CLS_DYNATCK
	EXPREL	1A13	0	15	CLS_DYNDEC
	EXPRANGE	1A14	20	100	CLS_DB1
	LEVEL	1A15	63	124	CLS_DB1OFF18
	CREST	1A16	0	8	CLS_CRESC
	DELAY	1A17	0	250	CLS_MS1
	LIMDLY	1A18	0	250	CLS_MS1
	SFTKNEE	1A19	0	1	CLS_ONOFF
	METERS	1A1A	0	5	CLS_MTRRES
	REF0DB	1A1B	64	100	CLS_DB1
Mid Section →	COMTHR	1A1C	13	124	CLS_DB1
	COMRATIO	1A1D	0	15	CLS_RATIO1
	COMATCK	1A1E	0	15	CLS_DYNATCK
	COMREL	1A1F	0	15	CLS_DYNDEC
	LIMTHR	1A20	76	100	CLS_DB1
	LIMRATIO	1A21	0	1	CLS_RATIO3
	LIMATCK	1A22	0	15	CLS_LIMATCK
	LIMREL	1A23	0	15	CLS_DYNDEC
	EXPTHR	1A24	2	100	CLS_CLS_DB1
	EXPRATIO	1A25	0	15	CLS_RATIO2
	EXPATCK	1A26	0	15	CLS_DYNATCK
	EXPREL	1A27	0	15	CLS_DYNDEC
	EXPRANGE	1A28	20	100	CLS_DB1
	LEVEL	1A29	63	124	CLS_DB1OFF18
	CREST	1A2A	0	8	CLS_CRESC
	DELAY	1A2B	0	250	CLS_MS1
	LIMDLY	1A2C	0	250	CLS_MS1
	SFTKNEE	1A2D	0	1	CLS_ONOFF
	METERS	1A2E	0	5	CLS_MTRRES
	REF0DB	1A2F	64	100	CLS_DB1
High Section →	COMTHR	1A30	13	124	CLS_DB1
	COMRATIO	1A31	0	15	CLS_RATIO1
	COMATCK	1A32	0	15	CLS_DYNATCK
	COMREL	1A33	0	15	CLS_DYNDEC
	LIMTHR	1A34	76	100	CLS_DB1
	LIMRATIO	1A35	0	1	CLS_RATIO3
	LIMATCK	1A36	0	15	CLS_LIMATCK
	LIMREL	1A37	0	15	CLS_DYNDEC
	EXPTHR	1A38	2	100	CLS_CLS_DB1
	EXPRATIO	1A39	0	15	CLS_RATIO2
	EXPATCK	1A3A	0	15	CLS_DYNATCK
	EXPREL	1A3B	0	15	CLS_DYNDEC
	EXPRANGE	1A3C	20	100	CLS_DB1
	LEVEL	1A3D	63	124	CLS_DB1OFF18
	CREST	1A3E	0	8	CLS_CRESC
	DELAY	1A3F	0	250	CLS_MS1
	LIMDLY	1A40	0	250	CLS_MS1
	SFTKNEE	1A41	0	1	CLS_ONOFF
	METERS	1A42	0	5	CLS_MTRRES
	REF0DB	1A43	64	100	CLS_DB1
Other Parameters →	PARLNK ♦	1A44	0	1	CLS_ONOFF
	NOMDELAY	1A45	0	250	CLS_MS1

LEV PAGE ♦	1A46	0	2	CLS_CLEVPG
COM PAGE ♦	1A47	0	6	CLS_CCOMP
LIM PAGE ♦	1A48	0	4	CLS_CLIMPG
EXP PAGE ♦	1A49	0	4	CLS_CEXPPG
LOW METER ♣	1A4A	-32767	32767	CLS_COMMTR
MID METER ♣	1A4B	-32767	32767	CLS_COMMTR
HIGH METER ♣	1A4C	-32767	32767	CLS_COMMTR
LGAIN ♣	1A4D	0	124	CLS_DBF1
MGAIN ♣	1A4E	0	124	CLS_DBF1
HGAIN ♣	1A4F	0	124	CLS_DBF1



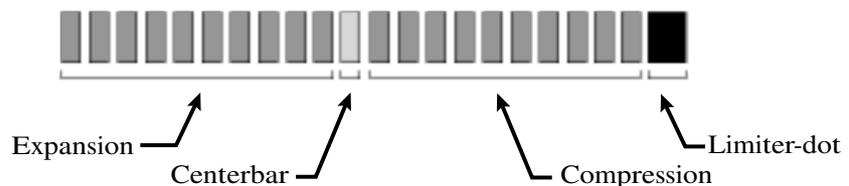
Parameters marked with a Fejl! Henvisningskilde ikke fundet. are only used for internal front-panel operations within the M5000 and have no effect on the audio-signal.

Parameters marked with a Fejl! Henvisningskilde ikke fundet. are read-only. Setting their value has no effect.

3.12.1 Meters

The Low, Mid and High meters indicate compression, expansion and limiting. The meters must be polled manually by your software. The M5000 makes sure that you always receive the highest value since the last poll (or lowest, if the expander is active), so all you have to do is poll the meters and display the bargraph. The M5000 is ready with new meter-settings 25 times a second, although you may choose to poll at a lower rate. It is recommended that you poll all 3 meters with the same request-packet in order to minimize MIDI-traffic.

The M5000 supplies you with information to display a meter that looks exactly like the meter on the M5000 display. The meter is 21 segments wide with a center at bar #11. Expansion causes the meter to move left, while compression causes the meter to move right. Limiting causes a dot to appear in the rightmost corner of the meter. Compression and expansion are mutually exclusive, and thus never happen simultaneously:



The M5000 scales the meters according to the METER-resolution for each band (eg. Par #1A1A for the low band). Don't confuse this parameter with the actual METER-readout (eg. 1A4A for the low band). If the meter-resolution is set to 5dB, then the compression section of the meter shows 5dB of compression, and the expansion section shows 5dB of expansion. See section 0 for a description of how the Meter-resolution parameters affect each other.

3.12.2 Meter Code-Example

The METER-readout parameter contains all necessary information on how to draw the meter. The limiter-dot is stored in the sign-bit (bit 15, if you've converted the parameter-value to a C-type short). You must first extract this bit and mask it off, before calculating the meters.

The compression meter moves in the range 0 up to 127, and the expansion meter moves in the range 256 down to 128. You should only display the first 10 segments in each range. An inactive meter is signified with a compression of 0.

The following demo-code gives a general outline of the decoding process:

```
#define  NOTHING -1

short   compression = NOTHING;
short   expansion = NOTHING;
short   limiting = FALSE;
short   masked;

// Parameter-value is passed in 'value'

if (value & 0x8000)
    limiting = TRUE;
masked = value & 0xff;
if (masked < 128)
{
    compression = masked;
    if (compression > 10) compression = 10;
}
else
{
    expansion = 256 - masked;
    if (expansion > 10) expansion = 10;
}
if (compression != NOTHING)
    // Draw compression
if (expansion != NOTHING)
    // Draw expansion
if (limiting)
    // Draw limiting
```

3.12.3 Mutual Dependencies

- LMXOVR must never be higher than MHXOVR.
- MHXOVR must never be lower than LMXOVR.
- METERS (Meter-resolution) for each band are hard-linked, meaning that they will always contain the same value. You should only choose to display and manipulate one of the parameters (fx. METERS - Low Band).
- ODBREF follows the same principle as METERS.

In each of the bands, the thresholds of the compressor and expander limit each other. The following criteria must be met for each of the bands:

- Compressor-threshold must never be lower than Expander-threshold.
- Expander-threshold must never be higher than Compressor-threshold.

3.13 TOOLBOX

Parameter-name	ID (Hex)	Min (Dec)	Max (Dec)	Class (Type)
MIX	1d00	0	100	CLS_PERCENT
INLEV	1d01	0	100	CLS_DB1
OUTLEV	1d02	0	100	CLS_DB1

INS-ON	1d03	0	1	CLS_ONOFF
BALANCE	1d04	-30	30	CLS_DBF1
MONO	1d05	0	100	CLS_PERCENT
LRSWAP	1d06	0	1	CLS_ONOFF
PHASE	1d07	0	2	CLS_LRPHASE
DITHER	1d08	0	3	CLS_DITHER
DITHER-TYPE	1d09	0	2	CLS_DITTYP
MS-IN	1d0a	-180	180	CLS_MSANGLE
MS-OUT	1d0b	-180	180	CLS_MSANGLE
FADECURVE	1d0c	0	1	CLS_MFC
FADER	1d0d	-80	0	CLS_DBF0
METER	1d0e	0	1	CLS_INOUT
RANGE	1d0f	0	2	CLS_MRNGE
TICKS	1d10	0	3	CLS_MTICKS
HOLD	1d11	0	2	CLS_MHOLD
LDELAY	1d12	0	3000	CLS_MS1
RDELAY	1d13	0	3000	CLS_MS1
EQTYPE1	1d14	0	3	CLS_EQTYPE
EQFREQ1	1d15	0	192	CLS_EQFREQ
PWIDTH1	1d16	0	16	CLS_WIDTH0
NWIDTH1	1d17	0	16	CLS_WIDTH1
SSLOPE1	1d18	0	3	CLS_SLOPE0
CSLOPE1	1d19	0	1	CLS_SLOPE1
PGAIN1	1d1a	-120	120	CLS_DBF1
NGAIN1	1d1b	0	100	CLS_DB1
SGAIN1	1d1c	-120	120	CLS_DBF1
EQON1	1d1d	0	1	CLS_ONOFF
EQTYPE2	1d1e	0	1	CLS_EQTYPE
EQFREQ2	1d1f	0	240	CLS_EQFREQ
PWIDTH2	1d20	0	16	CLS_WIDTH0
NWIDTH2	1d21	0	16	CLS_WIDTH1
PGAIN2	1d22	-120	120	CLS_DBF1
NGAIN2	1d23	0	100	CLS_DB1
EQON2	1d24	0	1	CLS_ONOFF
EQTYPE3	1d25	0	1	CLS_EQTYPE
EQFREQ3	1d26	0	240	CLS_EQFREQ
PWIDTH3	1d27	0	16	CLS_WIDTH0
NWIDTH3	1d28	0	16	CLS_WIDTH1
PGAIN3	1d29	-120	120	CLS_DBF1
NGAIN3	1d2a	0	100	CLS_DB1
EQON3	1d2b	0	1	CLS_ONOFF
EQTYPE4	1d2c	0	3	CLS_EQTYPE
EQFREQ4	1d2d	112	240	CLS_EQFREQ
PWIDTH4	1d2e	0	16	CLS_WIDTH0
NWIDTH4	1d2f	0	16	CLS_WIDTH1
SSLOPE4	1d30	0	3	CLS_SLOPE0
CSLOPE4	1d31	0	1	CLS_SLOPE1
PGAIN4	1d32	-120	120	CLS_DBF1
NGAIN4	1d33	0	100	CLS_DB1
SGAIN4	1d34	-120	120	CLS_DBF1
EQON4	1d35	0	1	CLS_ONOFF

♣ LPPM	1d36	-32768	32767	CLS_PPM72
♣ RPPM	1d37	-32768	32767	CLS_PPM72
♣ PHASE-CORR	1d38	-32768	32767	CLS_BAR72



Parameters marked with a ♣ are read-only. Setting their value has no effect.

If you wish to display the VU-meters, you should not use the local meters in this algorithm (LPPM and RPPM). A TOOLBOX algorithm has front-panel VU-meters as any other algorithm. If you use those, you only need to write one routine to display meters.

3.13.1 Special Considerations

Each of the bands in the TOOLBOX can be set to a number of equalizer-types. The Low and High bands have 4 selections, while the two Mid bands have only 2 selections. Each type of equalizer has it's own set of unique associated parameters. For example, a Shelve-type has a Slope-parameter, while a Parametric-type has a Width-parameter. As a product of this, the Low-band has 10 parameters, although not all of them are used at the same time. In the M5000, the display is swapped, to show only the relevant parameters. If the Parametric-type parameters are shown on screen, the Shelve-type parameters still exist as seperate identities, although they have no immediate function and aren't displayed. You will need to accomodate for this.

For each band, the EQTYPE, EQFREQ and EQON parameters stay fixed (meaning that they aren't swapped). The rest of the parameters for each band (Width/Slope and Level) are swapped. Please refer to the M5000 front-panel, to see how this works. If possible, you should retain this scheme as opposed to physically changing the type of objects in your software application.

4 System-Parameters

4.1 Overview

System-parameters are parameters just like algorithm-parameters except that they apply to the DSP-card as such or perhaps to the entire M5000 frame. Each slot has it's own standard set of parameters, such as Bypass, but care must be taken, because the parameters have the same numbers in all the slots. It is the slot# that distinguishes between the parameters, not the actual parameter-number.

4.2 System-Parameters

Parameter-name	ID (Hex)	Min (Dec)	Max (Dec)	Class (Type)
SYSMIXMODE	0100	0	2	CLS_MIXMODE
SYSBYPASS	0101	0	1	CLS_ONOFF
SYSGIN	0102	0	100	CLS_DB1
SYSCHANMODE	0103	0	3	CLS_CHNLMODE
SYSPHASE	0104	0	1	CLS_POSNEG
SYSIOMODE	0105	(See		CLS_IOMODE
♣ SYSCURRATE	0108	Section		
SYSMCLOCK	0109	0)		
SYSAIN	010a	-12	12	CLS_DBF0???
SYSAOUT	010b	-18	12	CLS_DBF0???
♣ SYSLOCKSTAT	010c	0	1	CLS_ONOFF
SYSMETERSHOW	0110	0	1	CLS_INOUT
SYSDOTYPE	0111	0	2	CLS_DOTYPE
SYSDOCPY	0112	0	2	CLS_DOCPY
SYSDADEMP	0113	0	1	CLS_ONOFF
SYSR68LEV	0114	0	1	CLS_ONOFF
SYSFSTTRIG	0115	0	1	CLS_ONOFF
SYSMETERL	0401			See section 0
SYSMETERR	0402			See section 0

Parameters marked with a ♣ are read-only.

4.3 VU-Meters

Even though the M5000 front-panel VU-meters only have 10 segments, the meter-information is in fact far more detailed. The SYSMETERL and SYSMETERR parameters contain the actual meter-readout in 1/8 dB steps, which is why steps must be taken to produce a useable meter.

4.3.1 Communication

The meters must be polled manually by your software at 1/25 second intervals. You may choose to poll at a lower rate, although a higher rate won't result in any improvement. It is recommended that you poll both the left- and right meters in the same request-packet in order to minimize MIDI-traffic.

The M5000 cannot guarantee to reply with the meters in the same packet, although the packets will be very close in time. For this reason, your software should be able to handle the left- and right channels separately.

4.3.2 Calculations

Some bits of SYSMETERL and SYSMETERR are reserved, so you should AND with 0x03ff before processing.

The meter-reply contains the actual meter position in 1/8 dB steps. A value of 0 means peak, while a value of 1 means -1/8 dB and so forth. If you wish to produce a meter that shows the amplitude in whole dB steps, simply divide the number by 8.

The DSP-clip flag is found in bit 13, which can be accessed by ANDing with 0x2000. This flag has a built-in timeout, so all you need to do is print it. The M5000 front-panel VU-meters use the 0dB LED to signal DSP-clipping, but this information should be printed separately wherever possible.

4.4 Hardware-Specific Parameter Ranges

Some parameters have ranges that depend on the physical DSP-card configuration.

5 Preset-Handling

5.1 Overview

The M5000 preset-handling facilities allow you to request information about presets (including edit-buffers) as well as transfer presets to/from the M5000. A convenient way of recalling presets via Sysex is also offered.

5.2 Conversions And IDs

A few new data-types are introduced for preset-transfers:

5.2.1 Preset-numbers

When a preset is referred to, it's number and bank is included in the Preset#. A Preset# is always spread over 2 bytes as an unsigned short and is combined/derived with the same methods as used for parameter-numbers.

In the C programming-language, the Preset# is calculated as follows:

```
presetNumber = number + (bank << 12);
```

where bank is one of the following:

```
0: Current Preset (Edit Buffer)
1: ROM
2: RAM
3: FILE
```

5.2.2 Algorithm-IDs

Each type of DSP-algorithm has a unique ID:

```
1 REVERB1
2 CHORUS
3 REVPITCH
4 REVERB2
5 NONLIN1
6 DELAY1
7 PITCH1
8 PITCH2
9 DELAY2
10 REVERB3
11 SAMPLER
12 AMBIENCE
13 TAPFAC1
14 DYNAMIC1
15 TOOLBOX
16 PAREQ
17 CORE
```

5.2.3 Preset-Names

Preset-names are always 8 ASCII-characters long with the unused character-places padded with spaces.

5.3 Preset Information

The following request-message allows you to request information about a preset or the edit-buffer:

```
Sysex-start  $f0
TC ID       $33
Device#     xx
Card#       Unused, unless edit-buffer is requested
Packet-type $03 - Request Preset Info
<Preset#>   Preset#
Sysex-end   $f7
```

The M5000 will reply with the following:

```
Sysex-start  $f0
TC ID       $33
Device#     xx
Card#       Unused, unless edit-buffer is replied
Packet-type $05 - Preset Info
<Name>      Preset-name
<Preset#>   Preset# (original Preset# if edit-buffer)
<Byte>      Algorithm-ID
<Byte>      Edited, 0=FALSE, 1=TRUE
Sysex-end   $f7
```

5.4 Recall Preset

This dump allows you to recall presets along the same path as the rest of your Sysex-communications:

```
Sysex-start  $f0
TC ID       $33
Device#     xx
Card#       xx
Packet-type $02 - Recall Preset
<Preset#>   Preset#
Sysex-end   $f7
```

Please note, that if the M5000 front-panel is showing the program-recall page, the Preset# will start flashing, because the preset you are in the process of recalling via the M5000 front-panel no longer is the current preset (because of this Sysex-dump).

6 C Programming-Interface

6.1 Overview

The C programming-interface is provided as a means to display the correct value for any given type of parameter. Without this interface, you would have to create all tables and conversions yourself in order to display the value of all parameters.

The interface consists of 2 files, CLASS.C and CLASS.H, which can be downloaded from the TC User-Club BBS. The phone-numbers are listed in the beginning of this manual.

6.2 Using The Interface

The interface is platform-independent and only requires the ANSI Standard Libraries `stdio`, `string` and `math` to be present.

The only routine you need to call is this:

```
void class_GetStr (char *s, WORD idClass, short v);
```

where `s` is an array of 7 `chars` to receive the string, `idClass` is the class-number as found in the algorithm-listings and `v` is the value.

Keep in mind that the string isn't automatically null-terminated. You can null-terminate the string by providing an 8-character string to `class_getStr` and then setting the 8th character to 0.

Because this code is taken directly from the M5000 application-software, the result is always printed as a 7-character string which is padded with spaces. You are free to modify the code to display the parameter-text in a less short-hand way, but keep in mind that the CLASS.C and CLASS.H files probably will be updated in the future to support new algorithms. Your CLASS.C and CLASS.H files will always remain compatible with existing algorithms, but if you would like to support new algorithms, you must either make all your modifications again or add the new parameter-types by hand.



Always make sure that the string you provide to `class_getStr` is large enough to contain the reply.

A few of the classes are irrelevant to most applications, but they have been left in the interface to simplify the process of providing it. You should not use classes such as `CLS_BAR72`, because their character-string reply requires custom-characters that are only available in the M5000.