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.

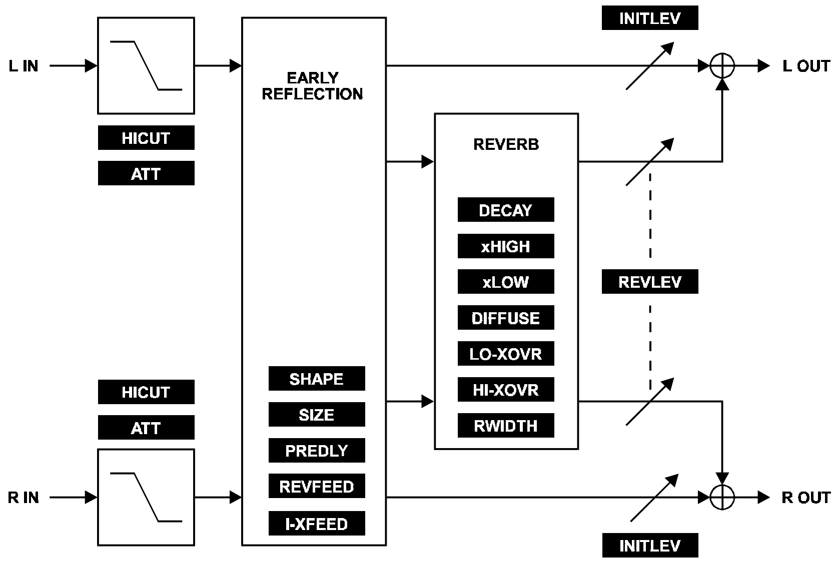
This section will explain the algorithms that come with the software version 2.0 They are as follows:

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 **Musik-vereinssaal, Austria.** Table 1 shows the actual sizes for the rooms simulated.

M5000 REVERB-1 & 2 algorithms										
								For the HALL pattern:		
									suggested	suggested
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 algoritm										

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).

[1]Only 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# 5IRAM-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.

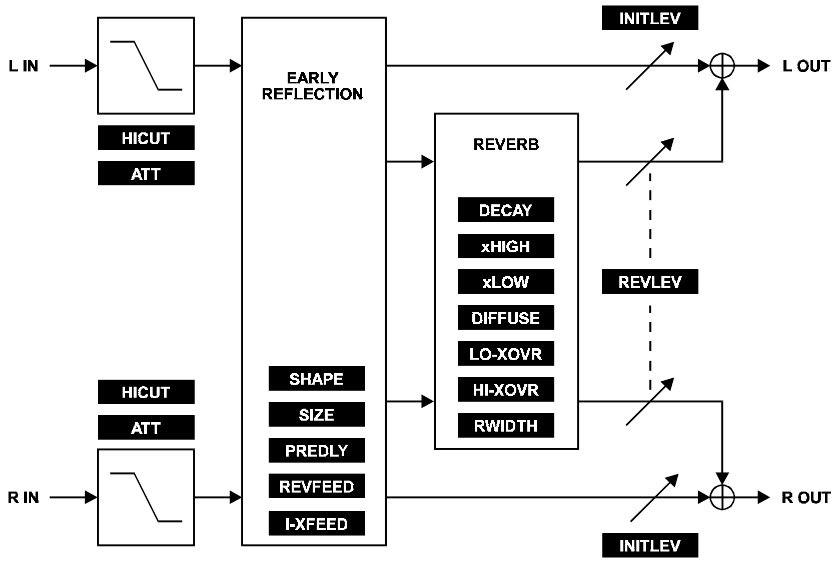
REVFEED — 0.0 - 100.0 mS; 0.0 - 300.0 mS[2] — Sets the time before the reverberating part of the signal starts to build up, relative to the early reflections PREDLY.

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

[2]Only 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

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		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 **Musik-vereinssaal, Austria.** Table 1 shows the actual sizes for the rooms simulated.

M5000 REVERB-1 & 2 algorithms										
								For the HALL pattern:		
									suggested	suggested
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 algoritm										

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

[3]Only 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# 5IRAM-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.

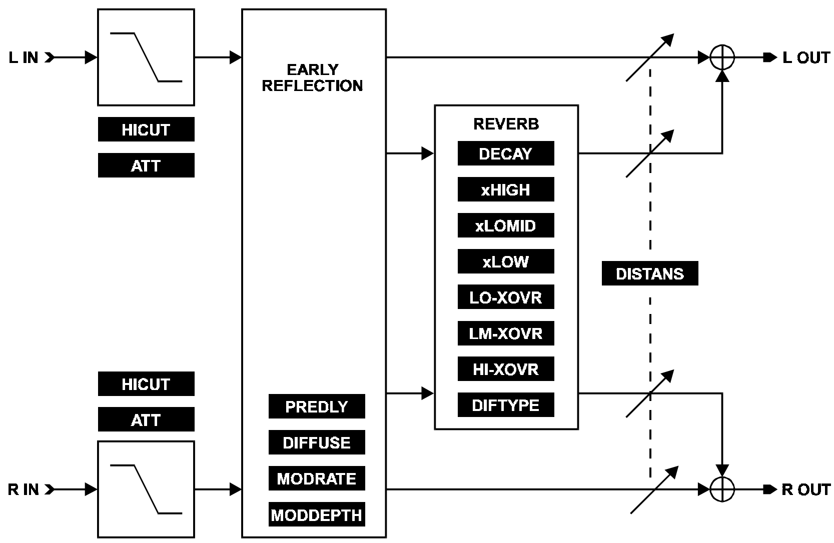
REVFEED	0.0 - 100.0 mS 0.0 - 300.0 mS[4]	Sets the time before the reverberating part of the signal starts to build up, relative to the early reflections PREDLY.
HICUT	500 Hz - flat	High cut filter, shelving type. Provides an overall reverb high frequency rolloff (6 dB per octave) that is well suited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

[4]Only 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.

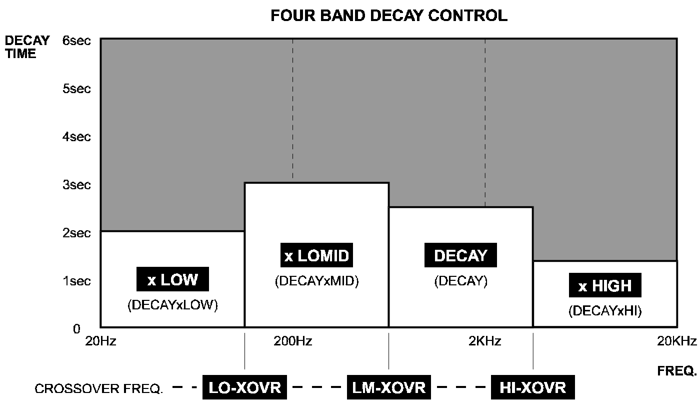
The diagram below is an addition to the signal flow diagram found in the "BASIC ALGO-RITHMS" module, page 2.



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.

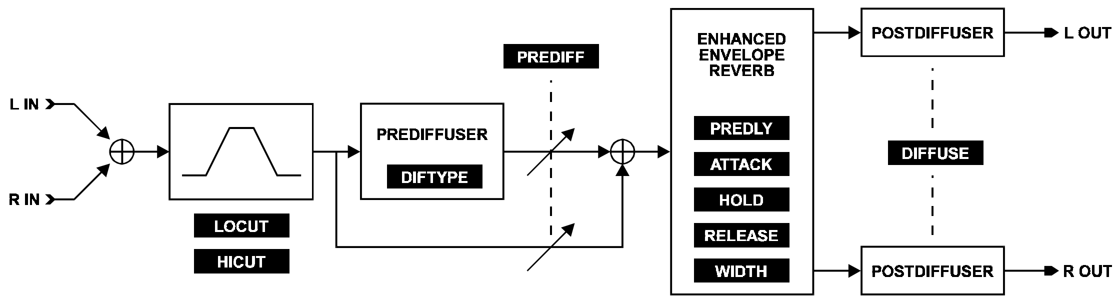


DIFFUSE	1 - 99	The DIFFUSE parameter simulates that the reflections in the room "hit" more or less uneven surfaces. With smooth walls low diffusion takes place. Walls that are uneven, with many angles, pockets or with dedicated diffusers cause the reflections to break into a high number of less identifiable 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

[5]Only 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 MODDPTH 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.
MODDPTH	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:

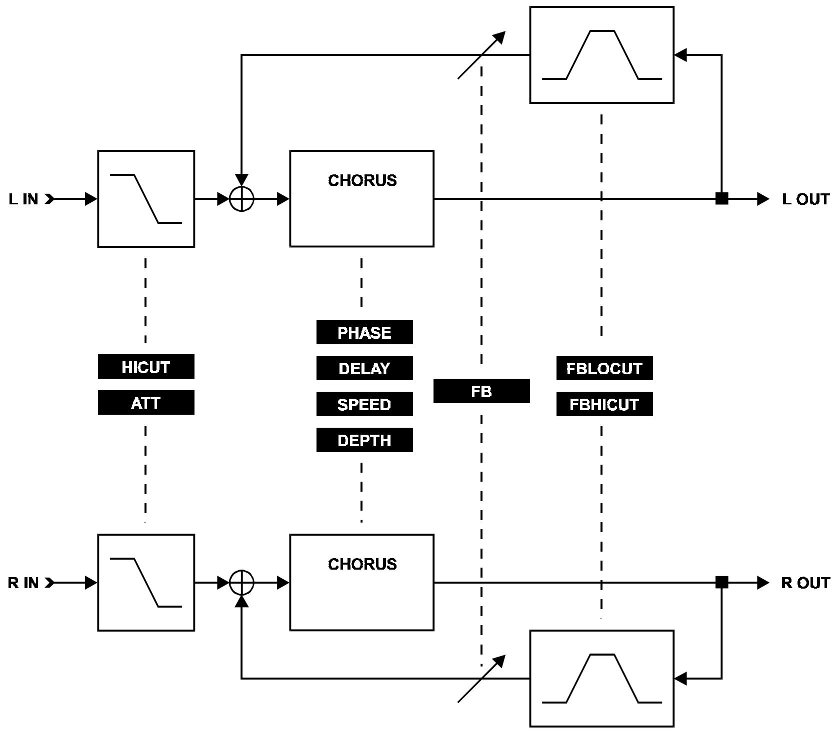
LOCUT	20 - 2.00 KHz	Low cut filter, shelving type. Provides an overall low frequency rolloff (6 dB per octave). Sets the cutoff frequency of the overall low cut filter in 1/3-octave steps.
HICUT	800 - flat	High cut filter, shelving type. Provides an overall high frequency rolloff (6 dB

		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 prediffussion 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-

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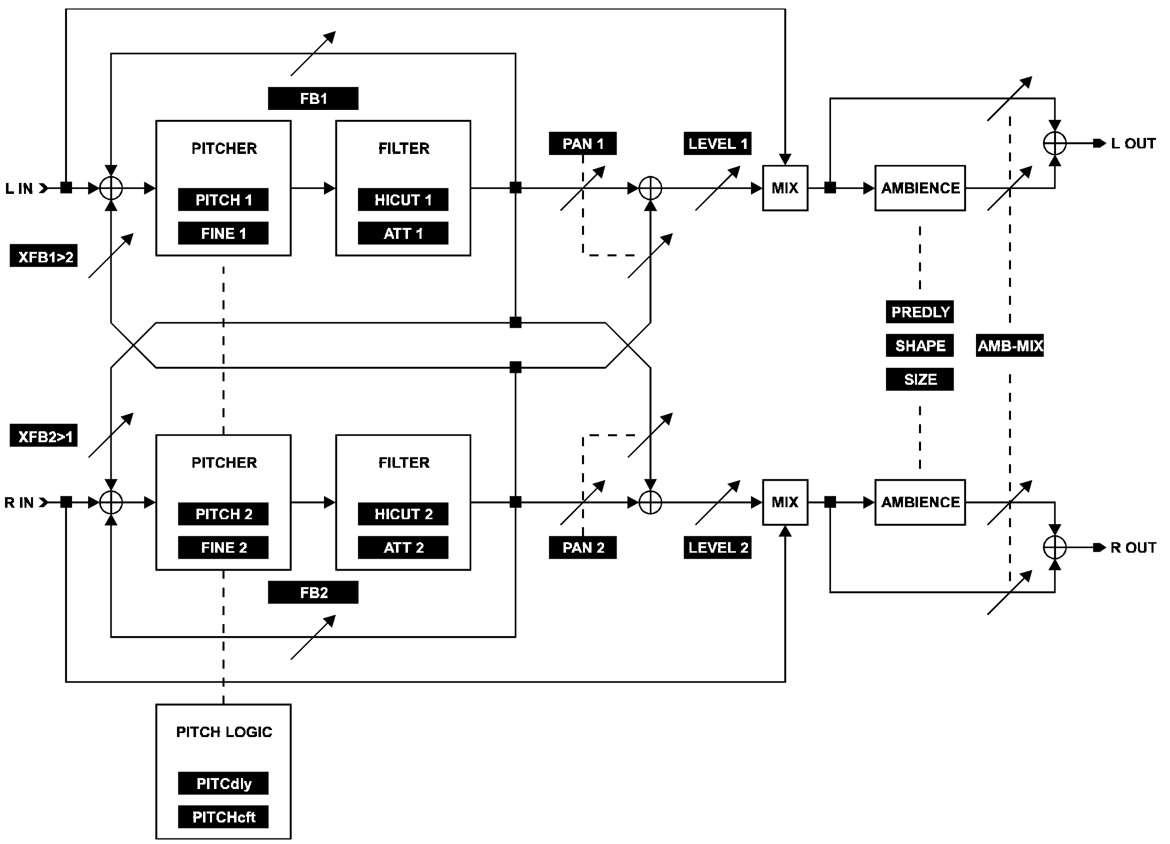
		menu under UTIL before setting this control. If the red overload LED still flashes, turn down INLEV a couple of dBs. The control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.
OUTLEV	off - 0.0 dB.	Sets the output level of the program in 0.5 dB steps. The function of this control is to maximize dynamic range by allowing the chorus algorithm to output maximum signal to the 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.
PHASE	0º - 90º - 180º	Determines the sine wave modulation phase shift between left and right channels. At 0º the left and right modulation will move in sync. At 180º the modulation will move the channels against each other.
DELAY	0 - 670 mS (Idx=32K) 0 - 1.360 mS (Idx=64K)	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.
FEEDBACK	0 - 99 %	Controls the amount of effect signal routed back to the chorus input (Flanging).
SPEED	0.1 Hz - 10 Hz	Controls the rate of sweep in a range from 1 sweep every 10 seconds to 10 sweeps every second.
DEPTH	0 - 100 %	Determines how wide a modulation (sweep) is produced.

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.

REVPITCH M5000

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 ambiance 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

		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

		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

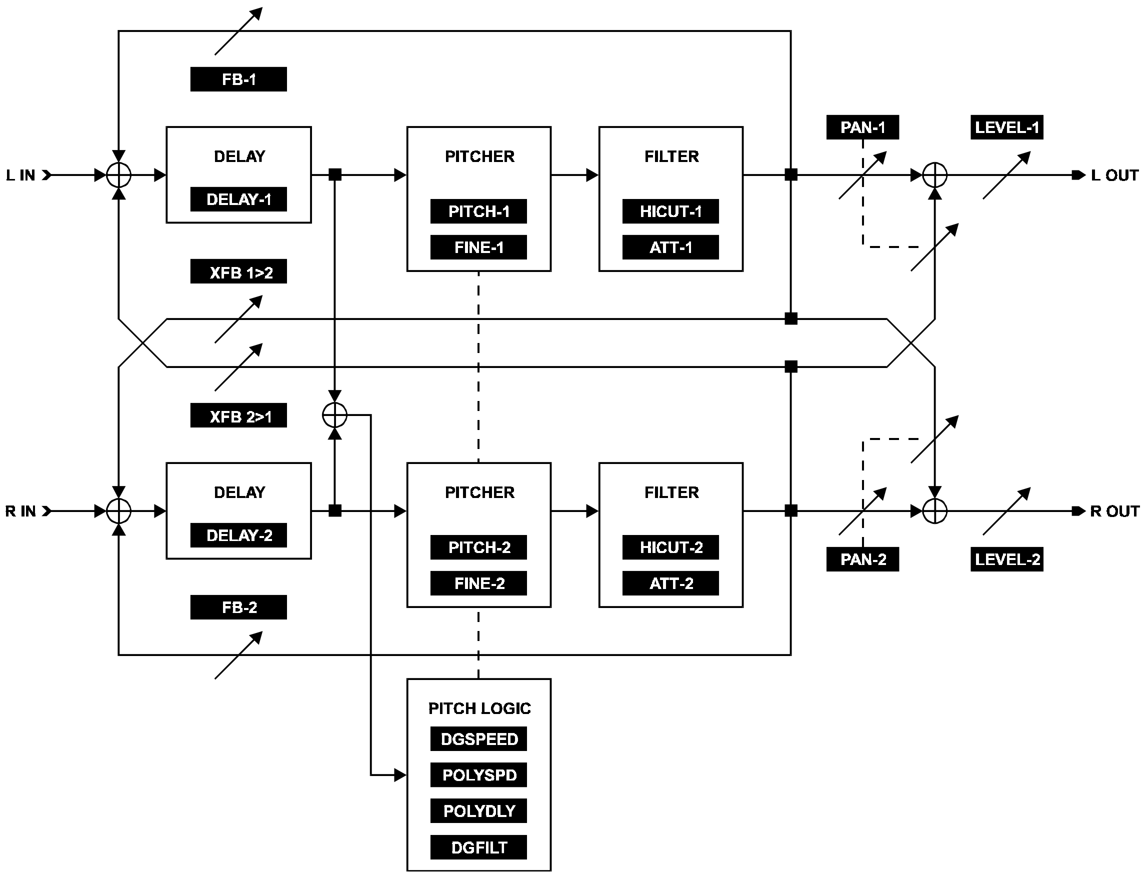
		for further description of the different shapes.
	DELAY	This is only one reflection (tab). With this shape it will act as a normal digital delay.
x SIZE	0.040 - 4.000	Scales the dimensions of the simulated space depending on the SHAPE chosen. A detailed description can be found under the REVERB-1 algorithm 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.

PITCH-1 M5000

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.

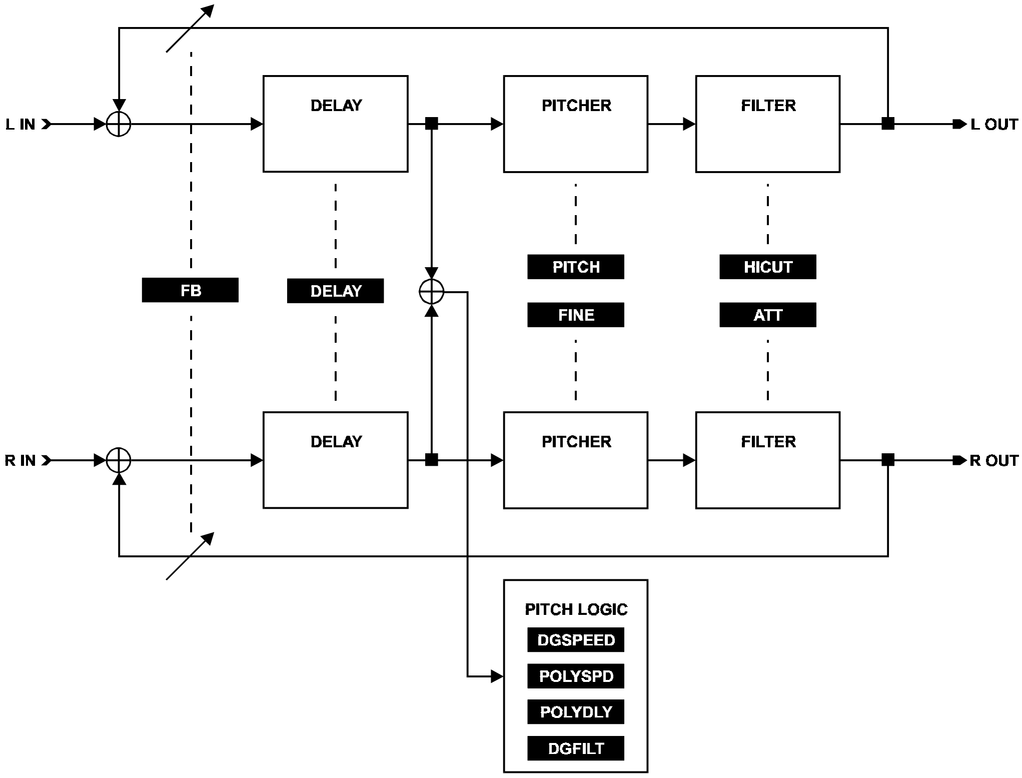
PAGE 4:

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

		ability to de-glitch 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 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.

PITCH-2 M5000

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.	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. The control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.

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.

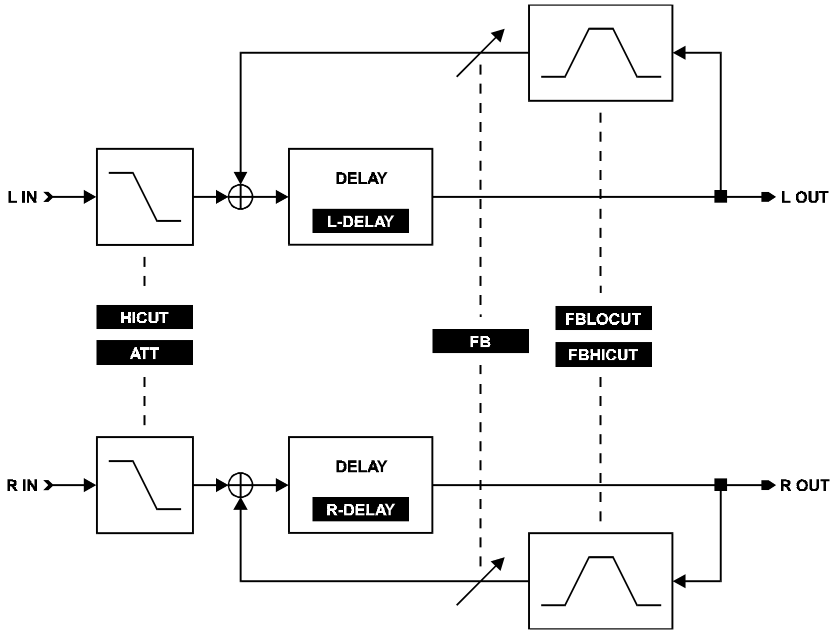
PAGE 4:

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 ALGO-RITHMS"-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

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		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 control. This control does not affect the bypassed signal level. Normally, you do not have to change the factory default setting.
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

[6]If 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).

		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 in 0.5 dB steps.

DELAY-2 M5000

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.

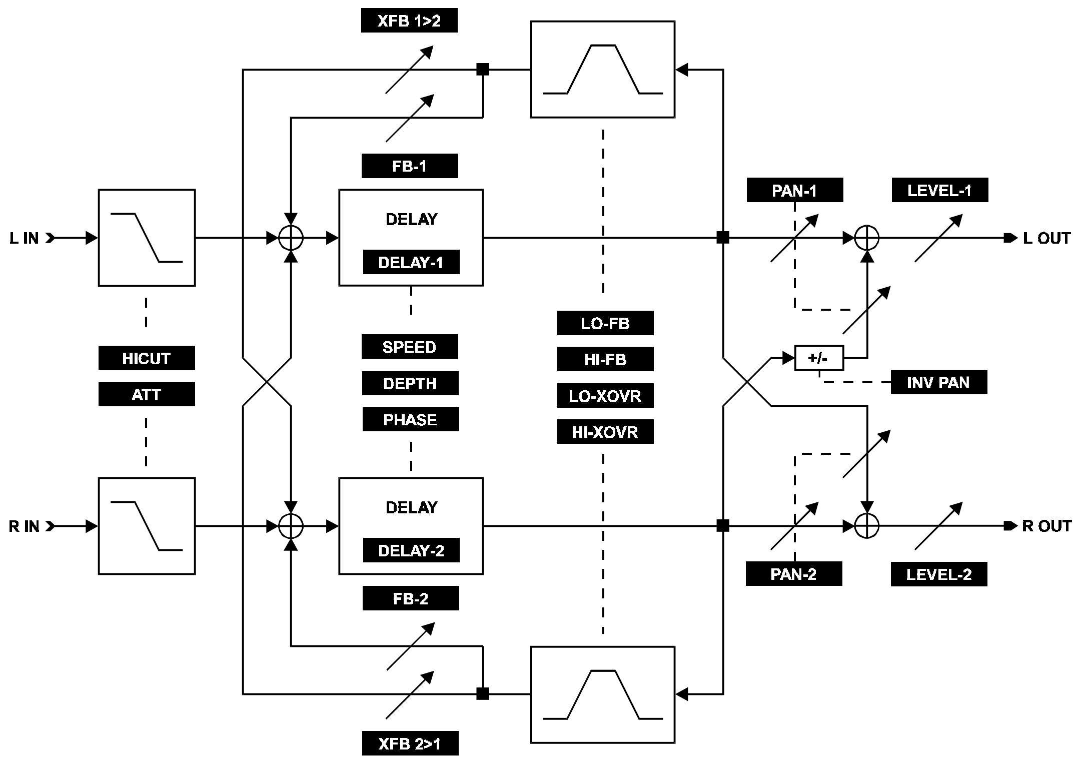


fig. 1

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.

[7] If 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.

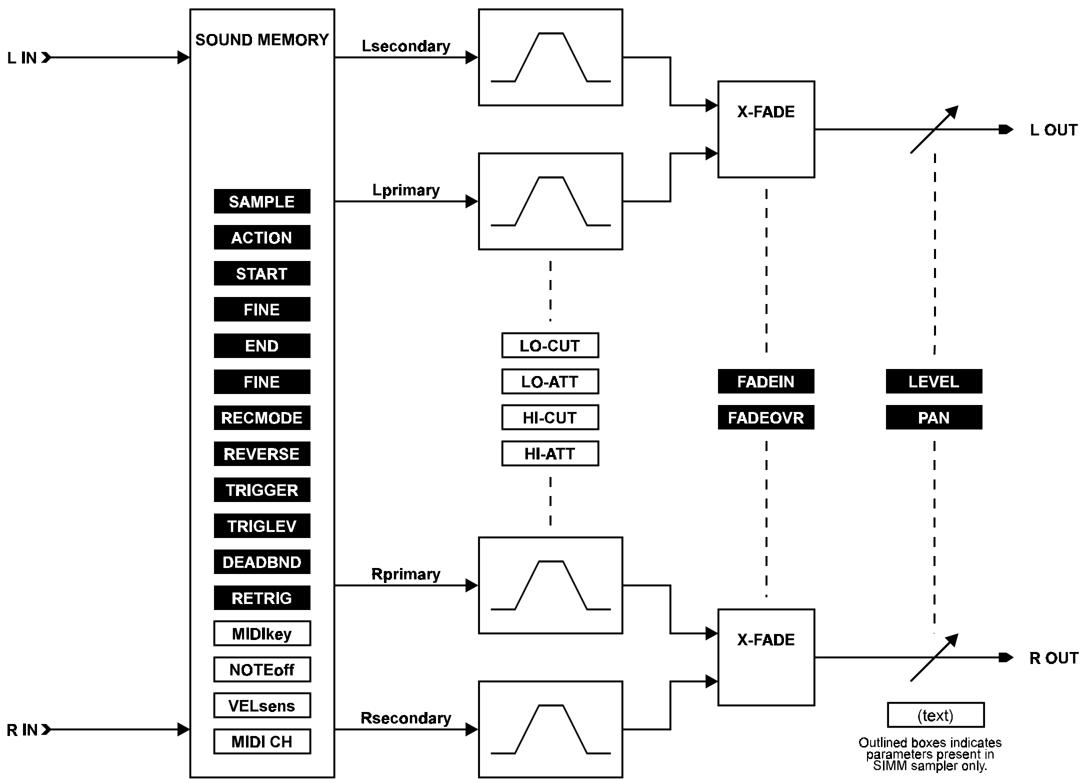
FB-1	-100 - 100%	The percent of positive phase and negative phase feedback for channel 1 (feedback path includes low-cut and hi-cut filters on PAGE 6).
FB-2	-100 - 100%	The percent of positive phase and negative phase feedback for channel 2 (feedback path includes low-cut and hi-cut filters on PAGE 6).
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.

SAMPLE-1 M5000

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

		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.
OUTLEV	off - 0.0 dB.	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		This parameter lets you decide your actions with the sampler:
	none	This is a "safe" mode. No action taken.
	play	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.
@ 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-

		time according to RECMODE and FREEMEM.
STATUS	ready?, armed!, playing, record., looping	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 (5IRAM) 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 retrig 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.

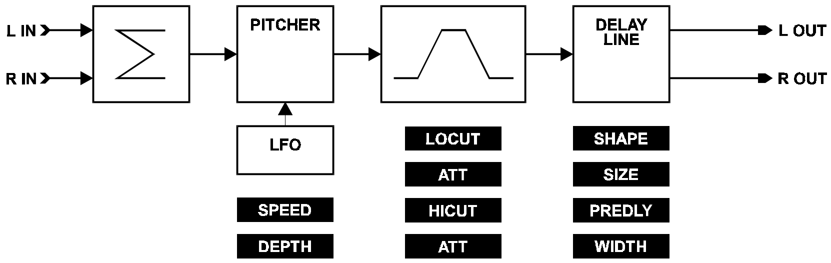
MIDI CH* omni - ch 1 - ch 16 Sets the MIDI channel on which the samples receives the MIDI commands. It must match the transmitting MIDI device. In **'omni'**-mode the samples will

* 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.

AMBIENCE M5000

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.

		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 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.
SHAPE		Room/Hall simulation/equivalent. With this control the initial pattern is chosen. Six distinctively different room shapes are available:
	HALL	The HALL reflection pattern is based on the acoustic properties of the **Boston Symphony Hall, USA.**
	FAN	The FAN pattern is based on the **La Scala Concert Hall in Milan, Italy.**
	PRISM	The PRISM pattern is from acoustic designers '**Golden Ratio**' shoe box shaped Hall.
	H.SHOE	The Horseshoe shaped pattern is based on the **Musikvereinssaal, Austria.**
	CLUB	The CLUB pattern is based on the typical dimensions of a club-sized location.
	SMALL	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.
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 in

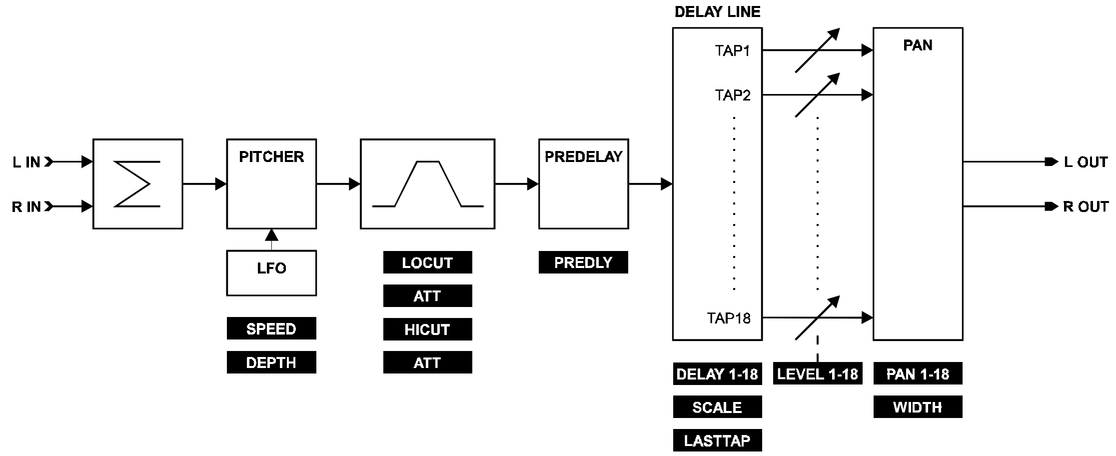
		REVERB-1 algorithm). Provided that the predelay setting is relatively short, the corresponding volume of the simulated space is changed radically with this control. For example; with the HALL initial pattern, the approximate room volume goes from 1.2 cubic meters to 1,280,000 cubic meters.
PREDLY	0.0 - 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.

TAPFAC M5000

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 you 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.

		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 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.

PAGE 2:

SCALE	0 - 100 %	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.
PREDLY	0.0 - 100.0 mS	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.
WIDTH	0 - 100 %	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.
LASTTAP	1 - 18	Selects the amount of active taps starting from 1. When set to 18 all 18 taps are active.

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:

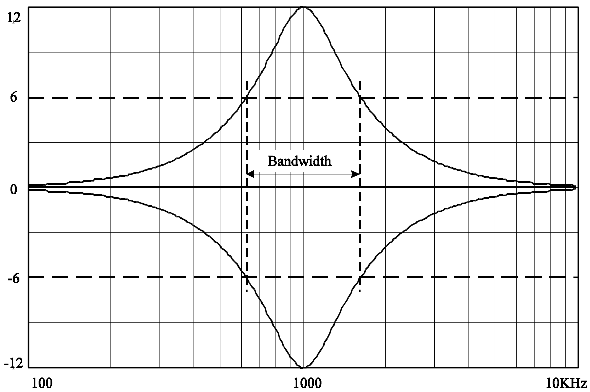


Fig.1 The bandwidth of the parametric EQ is expressed in octaves and is defined at half the EQ gain

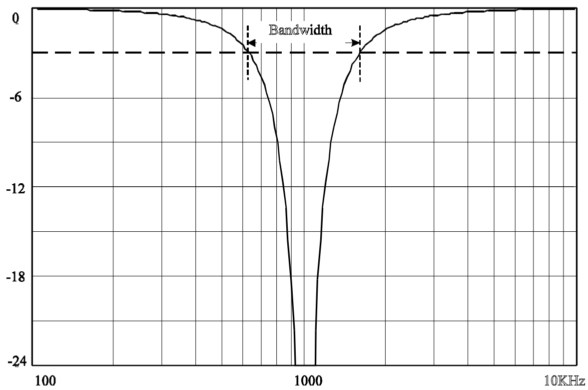


Fig.2 The bandwidth of the notch filter is defined at its -3dB points.

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.

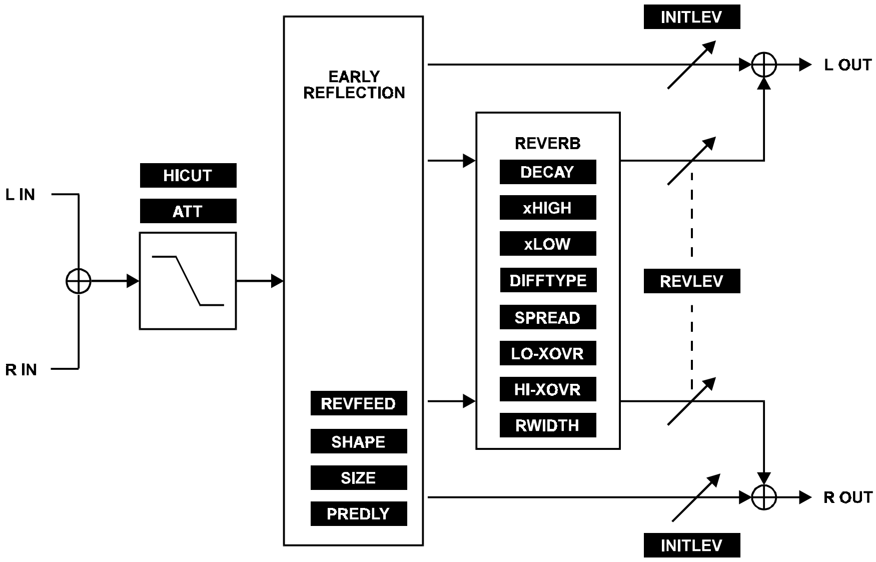
PAGE 6:

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 CORE[1]. 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.

[2]Only 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# 5IRAM-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.

[3]Only 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.

DYNAMIC1 M5000

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.

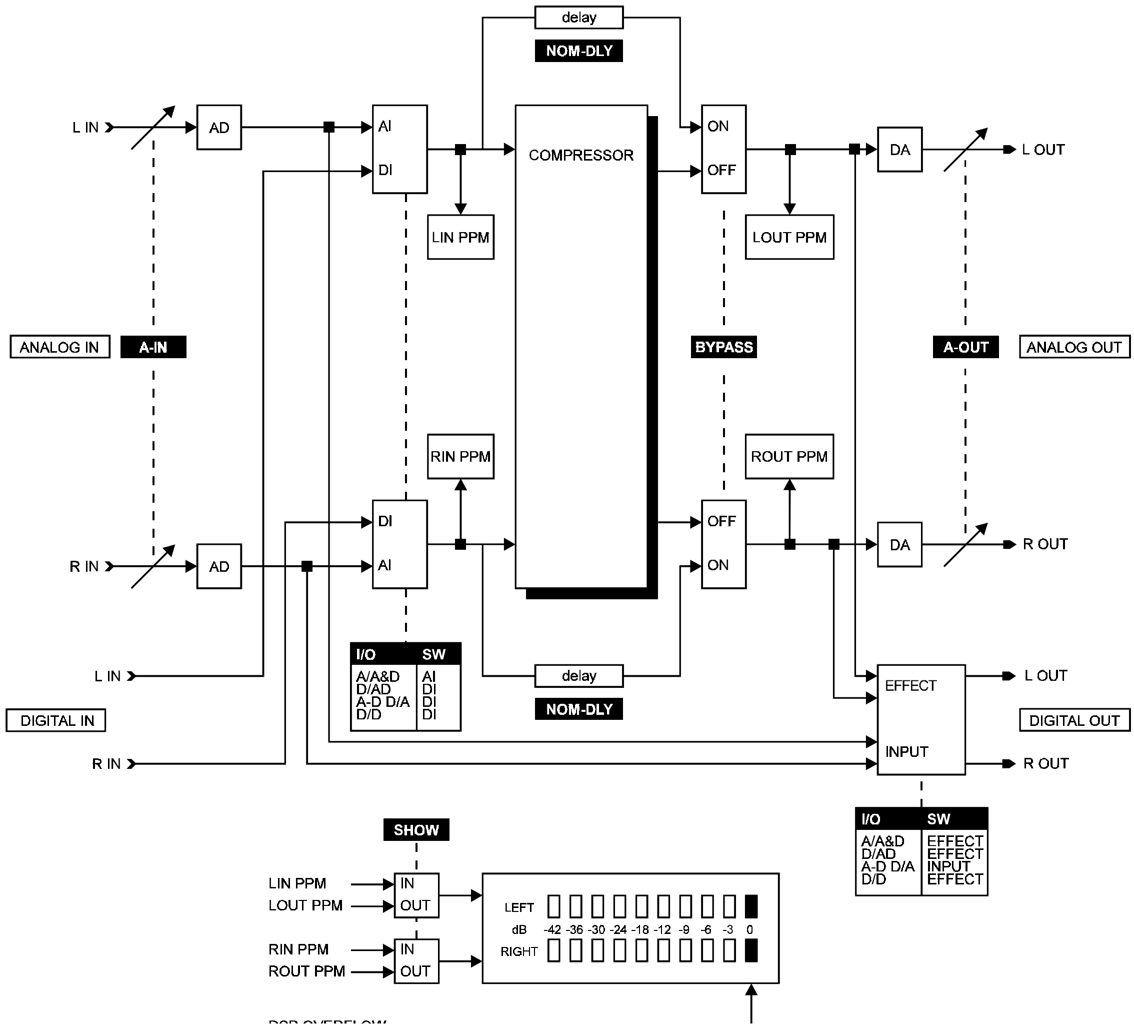


fig. 1

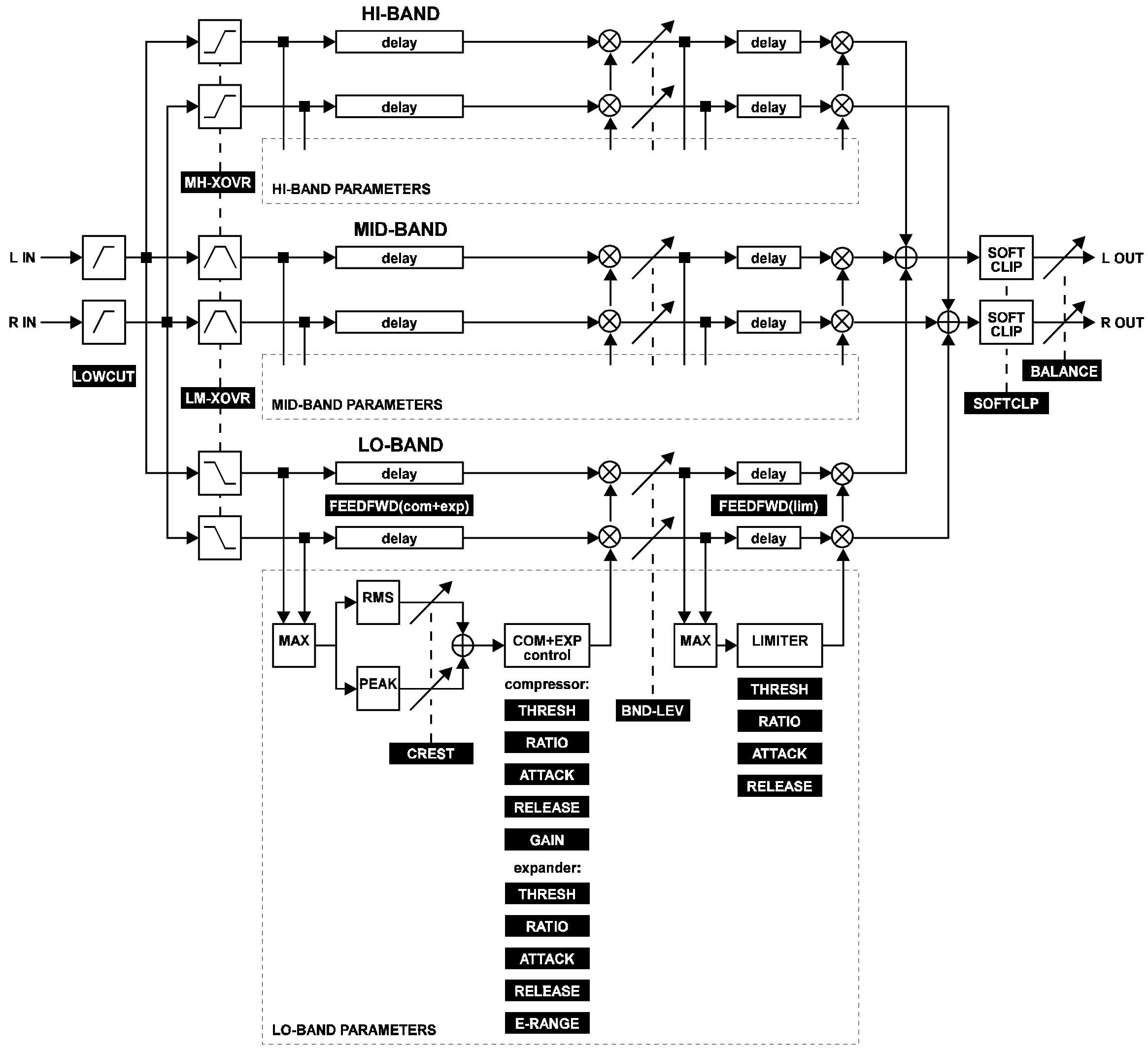


fig. 2

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

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		midband frequencies. When set to **low off** the algorithm is split in 2 bands.
MH-XOVR	mid off - 16.00 KHz	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.

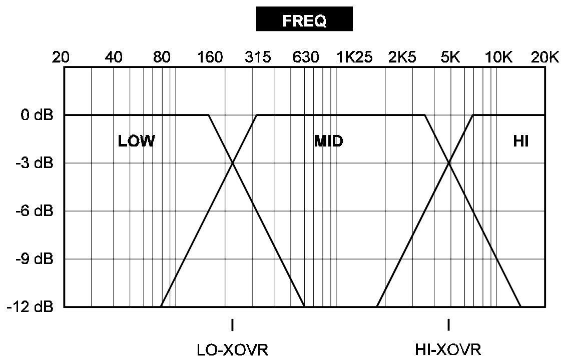

fig. 3

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.

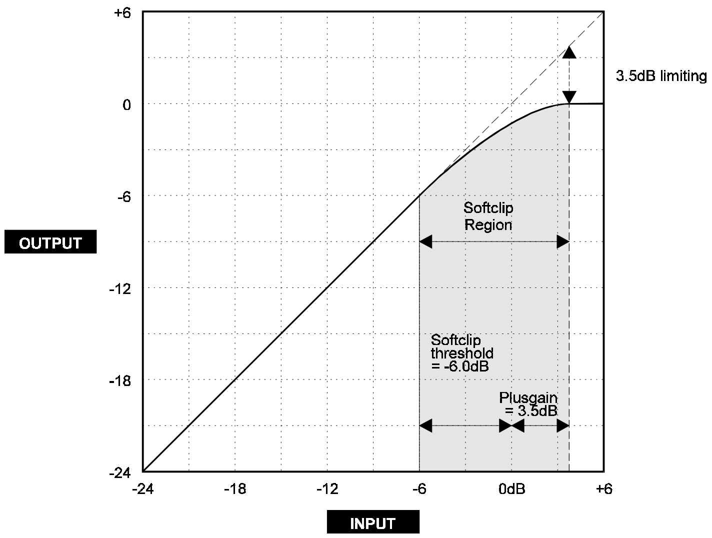

fig. 4

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.

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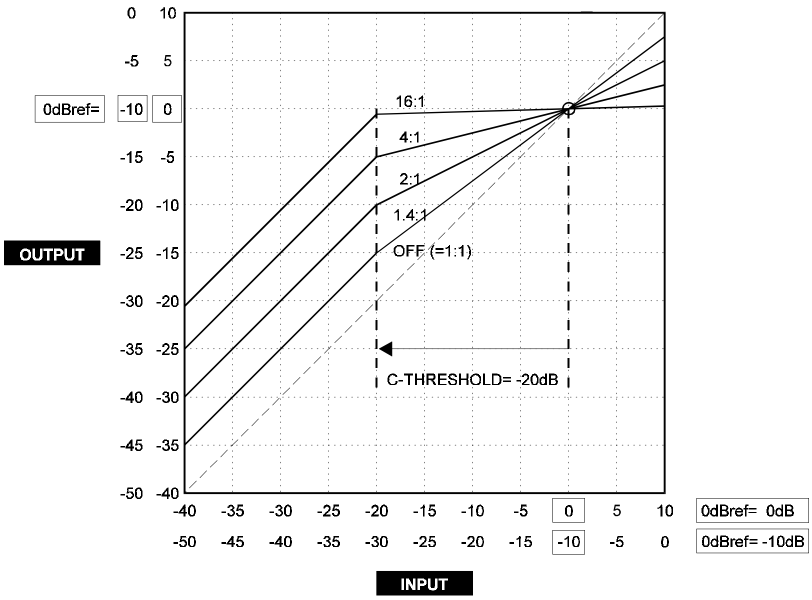


fig. 5

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……ˆƒ………M………‚………H………ˆ†…§

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 a 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 loose 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.

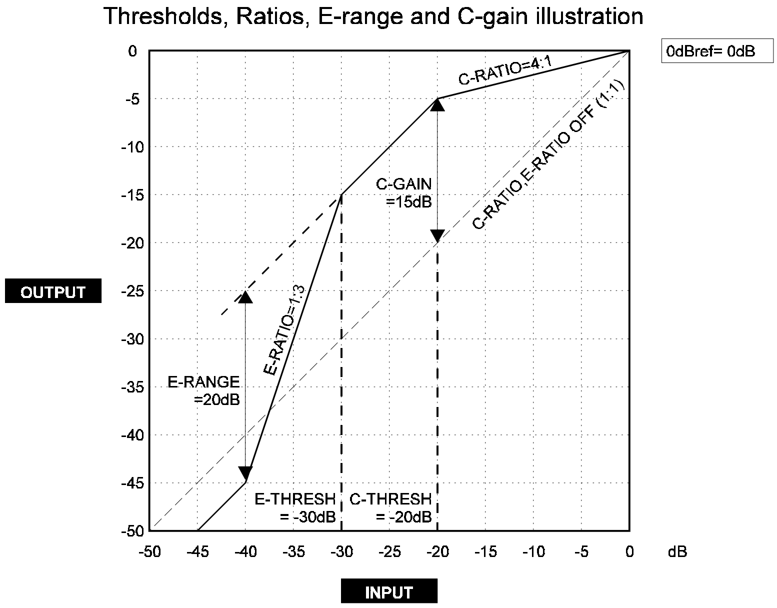


fig. 6

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 with behave as a standard analog compressor. NOTE: The overall NOM-DLY parameter (page 7) must always 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.)

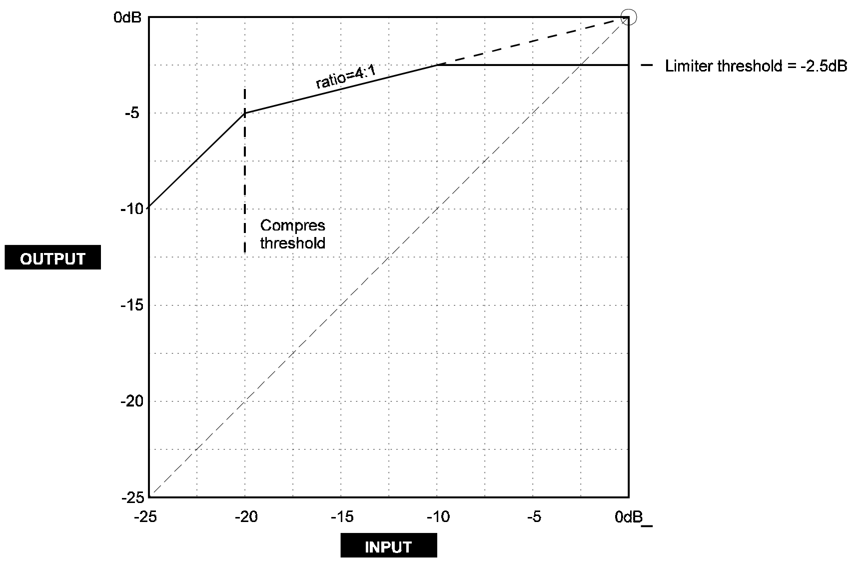


fig. 7

L-RATIO	off - infin:1	Gain reduction ratio.
L-ATTCK	30 µs - 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.

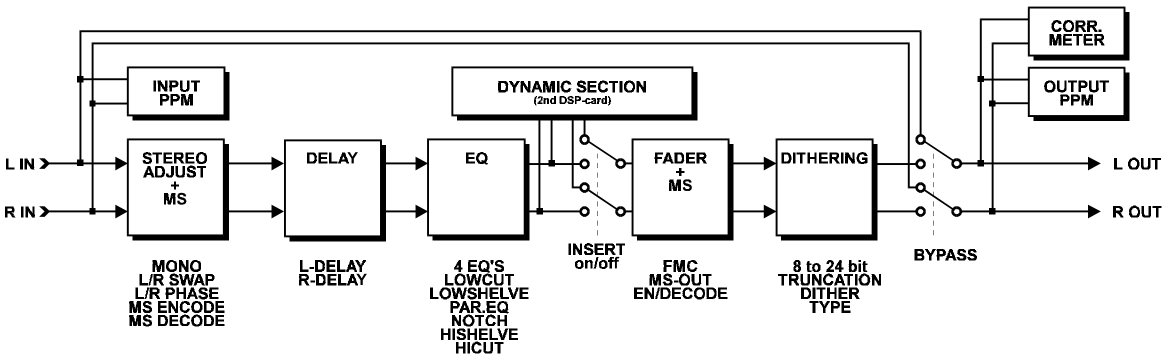
NOM-DLY	0.0 ms - 25 ms	Adjusts the nominal delay common to all bands. This acts a a DDL for the full audio spectrum. WARNING: The NOM-DLY should not be set lower than either of the FEEDFWD parameters found in the compressor or limiter pages as this parameter allows the FEEDFWD parameters to function 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.

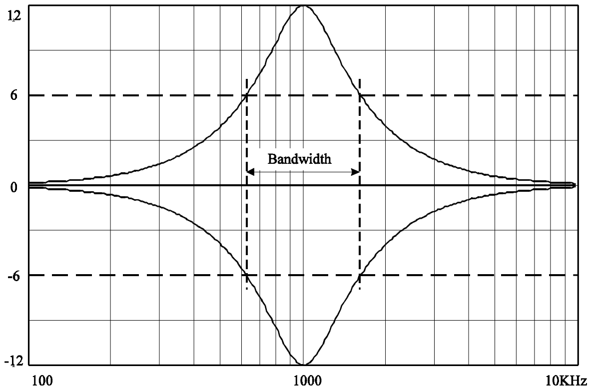


Fig.2 The bandwidth of the parametric EQ is expressed in octaves and is defined at half the eq gain

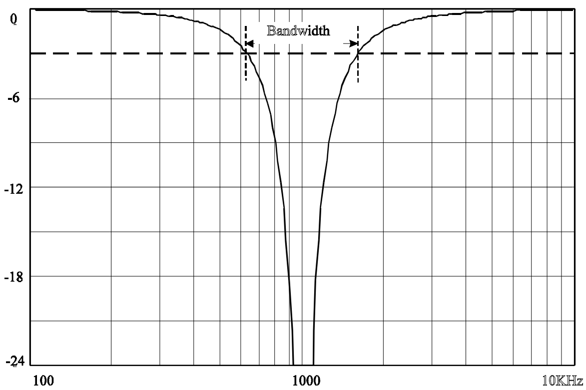


Fig.3 The bandwidth of the notch filter is defined at its -3dB points.

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 extend, 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 loose 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 to 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 is 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.

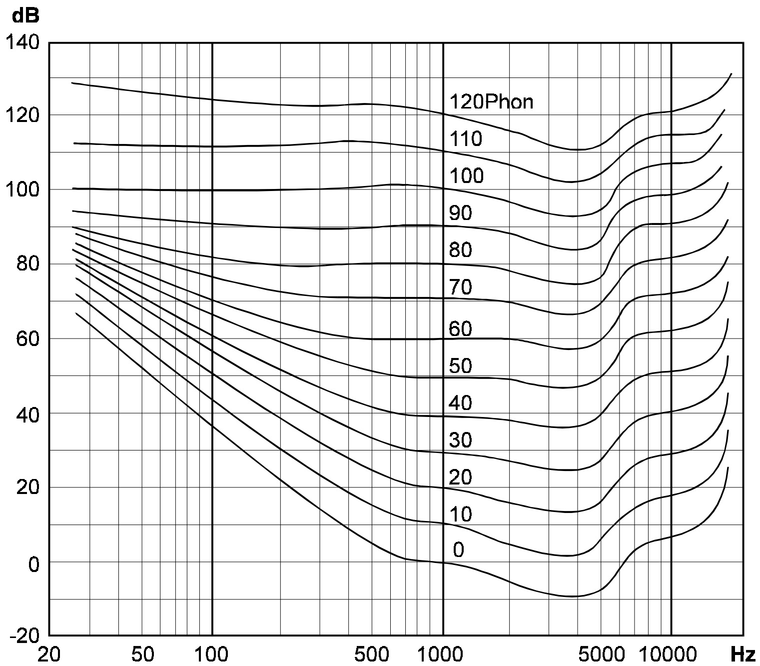


Fig. 4 The Fletcher-Munson Equal Loudness Contour Curves

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

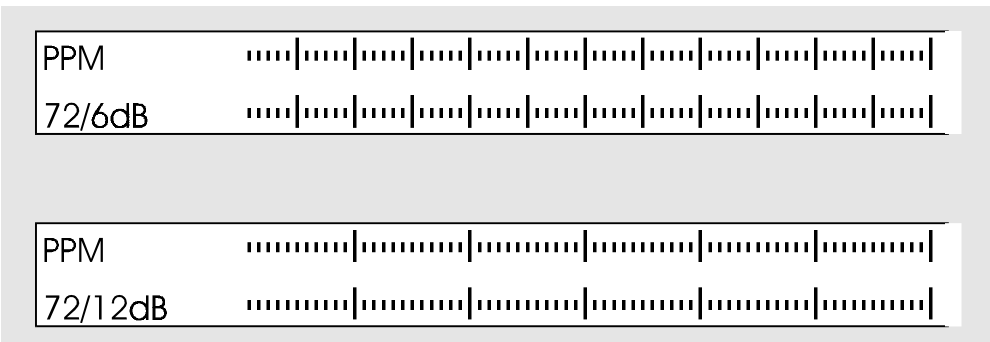


Fig. 5 The Meter range at 72 dB with the different ticks.

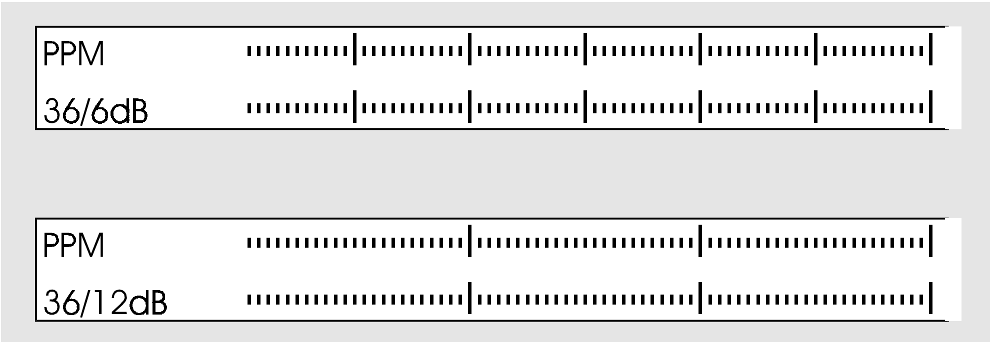


Fig. 6 The Meter range at 36 dB with the different ticks.

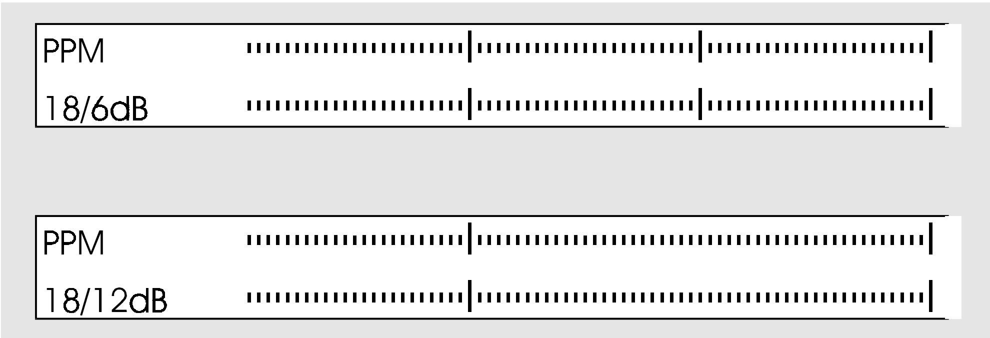
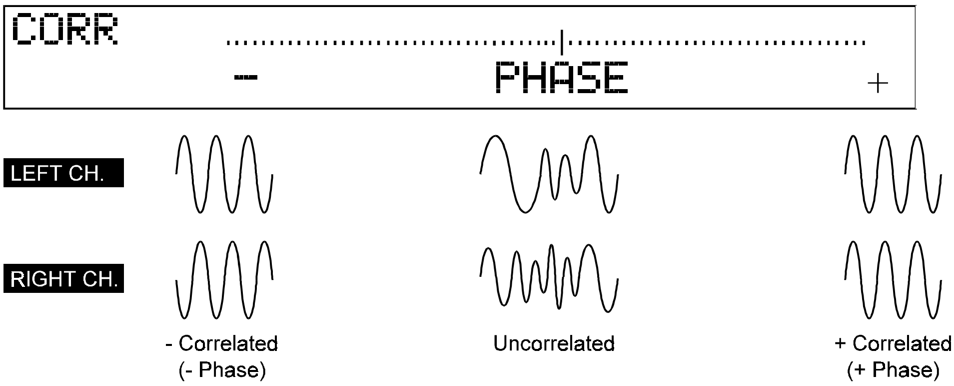


Fig. 7 The Meter range at 18 dB with the different ticks.

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.

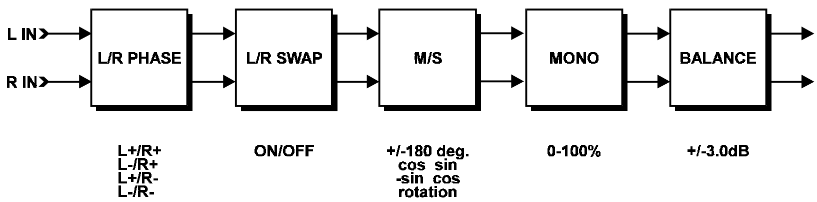


Fig. Signal flow of the input stereo adjustment parameters.

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.