digitalXtension

microWAVE PC

Programming manual

English

CE declaration

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TerraTec Electronic GmbH · Herrenpfad 38 · D-41334 Nettetal · Germany

hereby declare that the product:

digitalXtension microWAVE PC

to which this declaration refers is in compliance with the following standards or standardizing documents:

1. EN 55022
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This declaration is based on:

Test report(s) of the EMC testing laboratory

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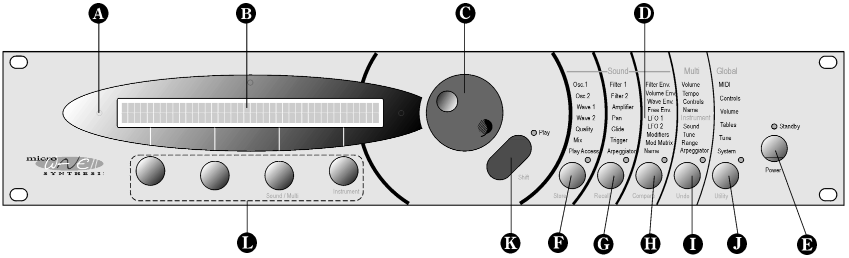
Control Features

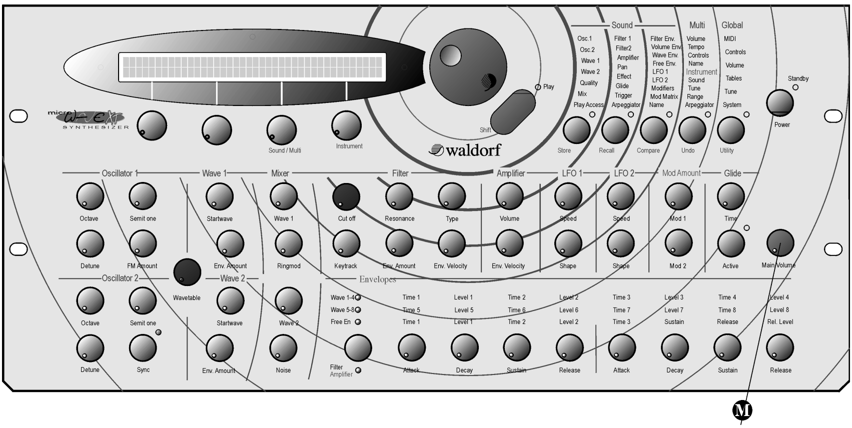
microWAVE PC „normal-mode“.



A	MIDI status LED	G	Select key for sound parameters
B	Display	H	Select key for sound parameters
C	Page dial for selecting sounds and parameter pages	I	Select key for multi-/instrument
D	Parameter pages	J	Select key for global parameters
E	Power switch with **Standby** LED	K	**Play** button for selecting the play mode
F	Select key for sound parameters	L	Value dials for adjusting parameters

Additional Controls of the microWAVE PC „XT-Mode“

In addition it offers individual controls for the most parameters. The items labeled on this page indicate special features that are available on the MicroWave XT only.



M **Main Volume** rotary control for setting the overall volume.

About this Manual.

This manual was written to help you become familiar with the microWAVE II/XT/PC. It will also help experienced users with routine tasks.

To avoid confusion, the terminology in this manual is based on the microWAVE II/XT/PC parameter names.

We also used a uniform set of symbols to alert you to topics of particular interest or significance.

Symbols.

Caution: The comments that follow this symbol will help you avoid errors and malfunctions.

☞ **Instructions:** Follow these guidelines to execute a desired function.

ⓘ **Info:** Additional information on a given topic.

Highlighted Control Features and Parameters

All of the microWAVE II/XT/PC's keys, pots and parameters are highlighted in **bold** letters throughout the manual. Also every control element has an unique position no. Ⓐ...Ⓜ which refers to the diagrams at the beginning of this manual. We suggest you make a copy of this page to have it at hand when necessary.

Example: • Press the **Play** key Ⓚ.

The microWAVE II/XT/PC's diverse modes and parameter pages are illustrated in a depiction of the display:

Octave 1	Semitone	Detune	Keytrack
-2	+07	+00	+100%

A given parameter's value range is indicated from low to high with the two values shown in ***italic*** letters, separated by three dots.

Example:

Semitone ***-12...+12***

Quick Start.

This chapter gives you a quick introduction into the microWAVE II/XT/PC and its features. It is written for those people that want to get a quick success without reading tons of manual stuff. Although the microWAVE II/XT/PC is a very complex device with many capabilities, its basic operation is quite easy to understand. But there are also more complicated things that make it necessary to take a deeper look into this manual from time to time.

Sound Mode

In Sound mode, the microWAVE II/XT/PC can play one sound at a time. You can select between 256 Sound programs, which are organised in two banks ***A001...B128*** and ***B001...B128***.

Selecting Sound Programs

1. Press the **Play** button Ⓚ to return to the program select page. The display now shows the program number and the name of the currently selected program (note: the program name and/or parameters can be different):

Play Sound AØØ1 Unisono WMF	Mode Sound	Main Vol. 1ØØ

   Play some notes on your MIDI keyboard. Listen to the sound.

2. If you want to adjust the microWAVE II/XT/PC's volume, use the rightmost value dial, labeled **Main Vol.**

3. Use the Page Dial Ⓒ to select other sound programs. Turning the dial clockwise increases the program number, turning the dial counterclockwise decreases it.

Editing Sound Parameters via Play Access

Now it is time to do some edits on a sound program. The easiest way for editing sound parameters is using the so-called **Play Access** page.

1. First, switch back to program ***A001***.

2. Press the **Play** button Ⓚ again to access this page. The display then shows 4 sound parameters that by adjusted directly via the corresponding value dials:

F1 Cutoff Ø92	F1 Reso ØØØ	F1 EnvAmt +29	FE Decay Ø84

3. Use the value dials to change the sound parameters and listen to the effect on the generated sound. Actually, you can define the parameter set in this page on your own. This is described later in the manual.

Comparing edited and original Program

You may always check your modifications against the original version of the program, though you can decide whether editing is going the right way or not.

1. Press the **Compare** key ***C*** on your PC keyboard.
2. The microWAVE II/XT/PC now uses the original parameter values as they were set before editing was applied. The display also shows these values. Play some notes to listed to the unedited sound.
3. Press the **Compare** key again. This brings you back to the edited sound program.

Recalling Edits

If you don't like the changed sound program, you can void the edits at any time and return to the original.

To do so, press the **Recall** key (or **R** on your PC keyboard).

Storing Programs

During the editing of a sound, this sound is available in the special microWAVE PC memory - the Edit-Buffer. Up to 8 Edit-Buffers can be set active at the same time, this means that up to 8 sounds can be edited simultaneously. If you try to edit a ninth sound, the former edited sounds will be set back to their default-values.(The former made changes will be lost.) By using the function "Store Sound" you save the edited contents of the Edit-Buffer to the chosen location in the microWAVE PC. When choosing another location, first save the instrument on the harddisk with the function "Save Instrument" from the "File"-Menu and upload this in a later stadium using the "Library", where you can select a location.

The function "Store-All Edits" saves all active Buffers at once.

Doing further Edits

We are now moving deeper into the sound editing capabilities of the microWAVE II/XT/PC. In the next steps we will show you how specific parameters act on the microWAVE II/XT/PC's behaviour. At first we like to play along with the filter.

1. Switch back to sound program ***A001***.
2. Press the second parameter select key **G**. The display changes to show the parameter page for Filter 1:

Cutoff	Resonance	Type	Keytrack
092	000	24dB LP	+050%

3. Use the first value dial to change the cutoff frequency of the filter, play some notes to hear the effect. Reduce the value to get a darker sound. Also change the **Resonance** setting. The sound gets a narrow character the more you turn up the control. Rise the setting to its maximum value. You will notice that an additional tone is generated.
4. After playing around a little, turn the **Cutoff** down to ***70*** and the **resonance** to ***20***. This should give you a good starting point for the next step.
5. Turn the Page Dial **C** clockwise to select the next parameter page. The display shows:

Cutoff Env. Amount	Env. Velocity Amount
+29	+00

6. Press a note on your keyboard and hold it down for a few seconds. You may notice, that the sound starts very bright but then gets darker more and more. This is the effect of the Filter Envelope that modulates the cutoff frequency. The modulation depth is controlled here by the **Cutoff Env. Amount** parameter.
7. Turn its setting down to ***0*** and look what happens: The sound starts in its dark state and no cutoff change can be heard.
8. Now set the value to a negative value, e.g. ***-10*** and press any note again. The sound then starts much darker than before and gets a little more brilliant after a while (you may raise the cutoff setting to get better results).
9. After playing around recall the original sound to get prepared for the next step.

Unisono mode.

This is a special feature of the microWAVE II/XT/PC that allows to use all voices for a single note. This makes the sound very fat. To show the difference to a normal sound, we are now going to turn the unisono mode off.

1. Use the Page Dial to go to the **Trigger 2** page. The page name is displayed in the upper right corner when turning the dial. The display shows:

Mode	Assign	Detune
Poly	unisono	Ø3Ø

2. Play some notes, then switch the **Assign** parameter to ***normal*** and listen what happens. The sound loses much of its power and fatness.

This needs a little bit of explanation: In normal mode, each note is played by one voice of the microWAVE II/XT/PC. This is fine for all situations when you want to play several notes, e.g. in a chord. In unisono mode, all voices are always used even for a single note. When you play two notes at a time, each one gets the half of the available voices. Use this mode especially for monophic lines. The Detune parameter is also very important in unisono mode. It determines how much each voice is detuned and therefore how fat the sound becomes.

3. Set the **Assign** parameter back to ***unisono***, if not already done.
4. Change the **Detune** parameter and listen to the effect. The detuning of the voices oscillators cause an audible sweep that is dependent on the parameter's value. The higher the setting, the stronger the sweep.
5. Set the **Assign** parameter to ***normal*** again. We will need this setting for the next steps.

Wavetables

They build the sound source from which everything derives. In this step we are going to change the sound program's wavetable.

1. To do so, press the first parameter select key ❻, then use the Page Dial ❸ to select the page.

Startwave	Phase	Wavetable W1
6Ø	free	Ø36 PulSync 1

2. Change the wavetable via the third value dial and play some notes. You may notice that the sound changes dramatically when moving from one wavetable to the next. Try to check out the following wavetables: ***014 Clipper, 021 Robotic, 028 FmntVocal, 054 Wavetrip2*** and ***060 Xmas Bell***.
3. After checking out the different wavetables, set the parameter back to the original wavetable ***036 PulSync 1***.

Ring modulation.

It is useful to add non-harmonic components to the sound that gives it a metallic character.

1. Use the page dial to select the **Mixer** page. The display now shows:

Wave 1	Wave 2	Ringmod	Noise
127	000	127	000

2. As you can see, the **Ringmod** parameter is already set to its maximum value. This is the reason why the basic sound character is so hard. Turn it down and play some notes. The sound gets much softer.
3. To understand what the ring modulation does, you should listen at its pure signal. Turn the level of **Wave 1** down to ***0*** and raise **Ringmod** to ***127*** again. Play some notes and listen to the result.

As you have seen in the **Mixer** page, the level of **Wave 2** is down at ***0***, which means that the whole sound is made upon one wave. We are now going to use the second wave, too.

1. Initially, turn the levels of **Wave 1** and **Ringmod** down to ***0***. You get a better impression what's going on.
2. Raise the value for the **Wave 2** parameter and play some notes. You will notice a total different "fall down" sound.
3. Mix in **Wave 1** again. Now both sound components are audible. Try to find a good balance for the levels.

Oscillators

The two waves are driven by two independent oscillators, that means they can have different pitch setting. Try out the following:

1. Use the page dial to select the **Osc 2 1** page. The display now shows:

Octave 2	Semitone	Detune	Keytrack
+0	+00	+06	+035%

Change the **Octave** setting and play some notes. Check out ***-2*** as a value.

The last thing we want to do in our little tour is to work with the envelopes. They determine the time characteristic of the sound program.

1. Select the **Filter Envelope** Envelope page. You must use the third selection key **H** to do this. The display shows:

FE Attack	Decay	Sustain	Release
000	084	000	070

2. Play some notes on the keyboard and decrease the **Decay** parameter. You will notice that the sound gets darker more quickly now.

3. Increase the **Attack** parameter. The effect you get is that the sound now starts dark and gets more brilliant. Finally it falls down to its dark state again.

To change the whole sound to a short and percussive hit, we have to use the Volume Envelope.

1. Select the **Volume Envelope** Envelope page. It is the next page after the Filter Envelope, so just turn the page dial one step clockwise. The display shows:

AE Attack	Decay	Sustain	Release
000	089	000	019

2. Decrease the setting of the **Decay** parameter. The whole sound gets shorter and shorter. At very low settings you will just hear a kind of click.

Multi Mode

In Multi mode, you can combine up to 8 sounds. Each sound in a Multi program is called an Instrument because it has some additional settings that belong to the Multi and therefore are not stored in the Sound program itself.

The are two main reasons for using a Multi program:

1. Using the microWAVE II/XT/PC with a sequencer. In that case you want to use several Sound programs at once, each assigned to a different MIDI channel.
2. Building layered sounds. By doing this you can get interesting combinations e.g. a chord sound that fades into a string pad.

Of course, you can use both methods in combination.

Selecting Multi Mode

The first thing we have to do is to switch from Sound to Multi mode.

1. Press the **Play** button Ⓚ to return to the program select page. The display now shows the program number and the name of the currently selected program:

Play Sound A001	Mode	Main Vol.
Unisono WMF	Sound	100

2. Turn the third value dial Ⓛ clockwise. The **Mode** setting changes from ***Sound*** to ***Multi***. The display now looks like this:

Play Multi 001	Mode	Main Vol.
MIDI Multi	Multi	100

3. Use the Page Dial Ⓒ to select other Multi programs. Turning the dial clockwise increases the program number, turning the dial counterclockwise decreases it.

Selecting Sound Programs for the Instruments

The next step is to select Sound programs for each instrument of the Multi.

1. Press the **Multi** key Ⓖ, to call the Multi/Instrument parameter pages. The display now shows the first page of the Multi parameters:

Multi Volume	
1ØØ	1

You can set the overall volume for the Multi program here. For now, leave it at its default value.

2. Use the Page Dial to select the Sound 1 page:

Bank	Sound Unisono	WMF
A	AØØ1	Inst. #1

3. Select a Sound program for Instrument 1 via the second value dial. In our example we select Program ***A018***. Play some notes on the keyboard to listen to the sound.

Bank	Sound Bigballs	DN
A	AØ18	Inst. #1

4. We are now selecting a Sound program for Instrument 2. You can switch between the Instruments via the fouth value dial. Turn the dial one step clockwise. The display shows:

Bank	Sound Unisono	WMF
A	AØØ1	Inst. #2

5. Select Sound program ***B003*** for the second Instrument. To change the Bank from ***A*** to ***B***, use the first value dial.

Bank	Sound Sqr Keys	WD
B	AØØ3	Inst. #2

6. To play Instrument 2, ensure that your master keyboard or sequencer is sending on MIDI channel 2. Play some notes on the keyboard.

You don't hear anything? Don't worry, everything went well. You have to activate the Instrument before it works as expected. As default, only Instrument 1 is active after initializing.

Activating the Instrument

Each Instrument has a **Status** parameter, where you can turn it on or off. This enables you to activate only those Instruments, that you really need.

1. Use the Page dial to select the **Sound 2** page:

Channel	Volume	Status	
02	100	off	Inst. #2

2. Change the **Status** setting to ***on***. Now the Instrument is active and you can listen to it when playing on the keyboard.

Building a layered Sound

Another exciting feature the Multi mode offeres is the capability to layer sounds. Such a layered sound consists of two or more Sound programs that are used in combination.

1. Select Instrument 3 and activate it as described above.
2. Choose a Sound program for the Instrument, e.g. ***A008 chaOSC***.
3. As expected, you can play the Sound program ***A008*** on MIDI channel 3. But this is not what we want to do here. In this case we want to combine it with Instrument 2 which is already setup.
4. The only thing you have to do is to change the MIDI receive channel of Instrument 3 in the Sound **2 page**. Use the first value dial to set it to 2:

Channel	Volume	Status	
02	100	on	Inst. #3

Both Instruments 2 and 3 now receive on MIDI channel 2. Therefore two Sound programs are played when you use this MIDI channel. You can layer more Instruments if you want.

Using an Instrument Arpeggiator

One of the outstanding features of the microWAVE II/XT/PC is its arpeggiator. In addition to the arpeggiator that can be used in a Sound program, each Instrument has an arpeggiator, too. That makes it possible to use arpeggios in a Multi program without editing any Sound program. You can even use the arpeggiator on Sound programs that normally don't use arpeggios.

1. Select the **Arpeggiator 1** page via the Page dial.
2. Select Instrument 2 via the fourth value dial. The display now shows:

Active	Clock	Range	
off	1/1	Ø1	Inst. #2

3. To activate the arpeggiator, change the **Active** parameter to *on*.
4. Now press and hold some keys on the keyboard. Make sure that is sends on MIDI channel 2 first.
5. You will notice that the sound changes every 2 seconds. This time period is determined mainly by two parameters: the **Clock** setting in the currently selected page and the **Multi Arpeggiator Tempo** in the **Tempo** page. Change the **Clock** setting to ***1/8*** and listen what happens: The arpeggio gets faster.
6. Play along with the other arpeggiator parameters and listen to the results.

That's okay for now. You have seen the basic things, but there is a lot of stuff left.

The best approach to the microWAVE II/XT/PC is learning by doing and so should you.

About Wavetable Synthesis

Basics

The sound generation of the microWAVE II/XT/PC is based on "the real" wavetable synthesis. This type of synthesis combines analog access and digital flexibility in a simple way. Although wavetable synthesis is a form of "sample playback" in principle, you should avoid this term because functionality, operation and results are totally different.

The ROM area of the microWAVE II/XT/PC consists of 64 wavetables, and the RAM area contains an additional 32 wavetables, which can be manipulated over MIDI via appropriate computer software.

A wavetable is a table made up of 64 columns. Each column represents one wave, that can be either located in the ROM or RAM area of the microWAVE II/XT/PC or calculated by an algorithm after selecting the wavetable. For the purpose of using a wavetable inside a sound program, it doesn't matter what source the wavetable comes from.

A wavetable itself contains no wave data, but is in fact a collection of up to 64 pointer entries referencing up to 64 waves. Not all columns of the wavetable have to contain entries. When one or several sequential columns contain no pointer, the microWAVE II/XT/PC calculates the waves for these locations automatically. The algorithm producing these "imaginary" waves uses an interpolation scheme that crossfades the "real" ones. E.g. when a wavetable cointains entries in column 1 and 5, the positions 2 to 4 are generated based on interpolation between the existing waves in column 1 and 5.

Please keep the terms "wavetable" and "wave" in mind and don't bring them into confusion.

Introduction

Wavetable synthesis gives the microWAVE II/XT/PC the unique sound character which makes it different from all other synthesizers and samplers. The principle of wavetable synthesis is not new, the PPG synthesizer "Wavecomputer 360", "Wave 2", "Wave 2.2" and "Wave 2.3" and also the Waldorf MicroWave (the first one) and Waldorf Wave use this concept. The microWAVE II/XT/PC contains some enhancements to wavetable synthesis which improve the sonic quality in a remarkable way.

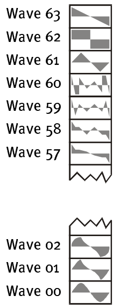
An introduction to wavetable synthesis needs some attention because its operation principle is different to other sound generating systems. Nevertheless you should spend a little time in understanding the basics, you will gain more than the effort it takes.

Please note that you cannot create your own wavetables or waves with the microWAVE II/XT/PC itself. To do so, you need a wavetable editor, a special computer program, that allows you to create and edit wavetables and waves. Please ask your local dealer for such an editor software. MicroEdit is not able to do so.

Overview

To illustrate the principle of wavetable synthesis, we start with an overview that is correct in a scientific way:

A wavetable is a table consisting of 64 waveforms. Each waveform is classified by its own very special sound character. Some wavetables contain waveforms with a similar sound character in between, others include waves with extremely different timbres. The following diagram shows a part of a wavetable.



You will notice, that the upper three entries in the wavetable consist of the classic analog type waveforms triangle, pulse and sawtooth. These three waves are identical in every wavetable. You can always use these classic synthesizer waves, independent of which wavetable is currently selected.

Both oscillators of a microWAVE II/XT/PC's voice use a common wavetable. However each oscillator can play a different waveform inside the table. E.g. oscillator 1 can play a sine wave from position 1 of the table while oscillator 2 is playing a sawtooth wave from position 63.

The main difference of wavetable synthesis compared to other sound generation principles is the facility not only to play one waveform per oscillator, but also to walk through the wavetable via different modulations. Therefore you can create wavetable sweeps. E.g. an oscillator can start with an sine wave and blend over to a sawtooth wave after some time. According to the wavetable used, the results can be very drastic - much more than any sample playback based system could ever produce. That is a unique feature of wavetable synthesis.

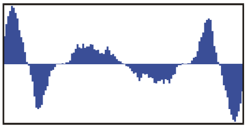
The capabilities of this principle are very strong. To give some examples:

- Each note on a 5 octave keyboard can access a different wave of the wavetable because such a keyboard has 61 keys, 3 less than the number of wavetable items.
- Different waves can be played depending on key velocity.
- An LFO can modulate the position inside the wavetable. Depending on the wavetable you can create subtle to drastic sound changes.
- Random controllers like e.g. the modwheel can change the position inside the wavetable. When you turn the wheel while playing a chord, each note's wave will be modified instantly.

These are just a few examples of the capabilities the microWAVE II/XT/PC's wavetable synthesis offers. In the following paragraphs we move deeper into the subject, and by the way we get a little more specific.

Wave

A wave is the digitally stored image of a single wave cycle. From this point of view a wave is identical to a sample that is looped exactly after one cycle. The difference to a sampler or ROM sample player is that all waves have the same length and they are played at the same pitch. A typical wave looks like this:



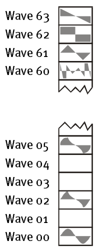
The diagram shows the symmetry of the waveform which is mirrored in its middle. In fact most waves in the microWAVE II/XT/PC are made up in this way so that only the first half of the cycle is stored in memory and the microWAVE II/XT/PC calculates the missing part on its own. At this point we see one extension to the classic PPG systems and the first MicroWave: The microWAVE II/XT/PC can also store whole wave cycles. This feature becomes interesting in all those cases where analog-type waveforms with different pulse width or additive created waveforms with different phase shifts of the harmonics should be generated. These sophisticated timbres were especially not realizable with the first generation wavetable synthesizers.

Wavetable

In fact a wavetable does not consist of waves but of pointers to them. The microWAVE II/XT/PC stores wavetables and waves separately, numbered from 001...096 for the wavetables and 100...600 for the waves.

In a wavetable up to 64 of these pointers are combined, each pointing at one of the 500 waves. The term "up to 64" means that a wavetable can contain even less pointers. In this case the missing entries are filled automatically by the microWAVE II/XT/PC as soon as the wavetable is selected. At least 5 pointers must be present in every wavetable, one at the first position and 4 at the last. Three of the four positions represent - as already described above - the classic synthesizer waveforms triangle, pulse and sawtooth.

E.g. the wavetable shown below contains pointers to waves at positions 00, 02, 05, 60 plus the three classic waves at positions ***61...63*** (we will ignore these three last ones for now).



Now imagine an oscillator sweeping through these wavetable to play one of the waves.

- When position ***00*** is selected, the oscillator plays the wave referenced by the wavetable.
- When position ***01*** is selected, the oscillator plays a wave which is calculated by the microWAVE II/XT/PC without being stored in memory directly. The shape of this wave is interpolated between the shapes of the previous and the next existing wave, both mixed with different amplitude settings. In the given example a wave with an amplitude relation of 50% to 50% from the waves on position ***00*** and ***02*** would be the result.
- When position ***02*** is selected, the microWAVE II/XT/PC plays a "real" wave again, the one referenced by the table position.
- Position ***03*** and ***04*** work similarly to position ***01***. Again, the waves to be played are calculated by the MicroWave. In this case the gap is bigger because two positions in the wavetable are empty. As a result a wave mix of 2/3 to 1/3 (i.e. approx. 66% to 33%) is generated for wave position ***03***. As you can see, the previous existing wave is more weighted here. At position 04 the calculation works vice versa, i.e. 1/3 of wave ***02*** amplitude and 2/3 of wave 05 amplitude.
- On position ***05*** a stored wave is played again.

If the oscillator would move up and down between positions ***02*** and ***05***, a continious change of the timbre would be noticed. It is a little bit oversized to call this "continuous" when not more than 4 positions are available but imagine no further wave pointers are stored between position ***05*** and ***60***. Then you will get a very smooth timbre change by moving from position ***05*** to ***60***.

And what about hard timbre changes? Now take a look at the classic waveforms on positions ***61...63***. As there are not any blank positions between these waves the resulting timbre changes are very hard.

What else can we do?

In addition to the described structure, the microWAVE II/XT/PC can generate wavetables and their corresponding waves via mathematical calculations. Such wavetables are called "algorithmic wavetables". The speciality about these wavetables is that they don't need any real waves to generate interesting timbre changes.

E.g. the calculation scheme for an algorithmic wavetable can be as follows: Take a pulse wave for position 00 and remove the last samples for every step, so that a single sample remains on position 60. The result is a wavetable with pulse waves of different pulsewidth.

The different base algorithms for such wavetables are:

- synchronisation
- pulse width modulation
- FM
- waveshaping

Summary

You should keep the following sentence in mind because it describes the essentials of the wavetable synthesis:

ⓘ A wavetable is a table of pointers to up to 64 waves, in between you can move randomly.

Creating own Wavetables

Sooner or later you want to create your own wavetables and waves maybe with corresponding 3rd party software..

Therefore we would like give you a short introduction into the basics of creating wavetables.

The biggest part of the microWAVE II/XT/PC's wavetables contain between 8 and 16 waves, some of them consist of fewer, some have more. As you can see, you don't need to fill all positions of a wavetable with waves to get interesting sweeps. Take your wavetable editor and look into some of the ROM wavetables. E.g. wavetable 01 is made up of very few waves while wavetable 28 contains a lot of them.

When you want to create a wavetable that simply fades from a pulse wave to a sawtooth waveform, you need exactly two waves. The first one, a pulse wave, on position 00 and the second one, a sawtooth wave, on position 60.

Look into the ROM waves. Consider these waves as a big collection for your own wavetables. E.g. you will find a sawtooth, a pulse, a triangle and a sine wave already there. So you can construct a whole new wavetable out of the ROM waves.

History

At the end of 1970, Wolfgang Palm, the founder of PPG, had the idea of recreating the sound and behaviour of analog circuitries through a digital representation of oscillator waveforms with different filter settings. He then stored these waveforms sequentially into a so-called wavetable and added features to scan through this wavetable by envelope, LFO and the like. The result was a sound that changed its timbre without using any kind of analog filtering or other processing like FM or ring modulation. These individual timbre changes that were different from anything else known at that time made up the typical "wave sound". The first synthesizers built in the early 80s that used this technique were the PPG 340/380 - Wave Computer and the PPG 360 Wave Computer. Both models yet without analog filters.

Wolfgang Düren, responsible for the distribution of the PPG synthesizers at that time, was able to convince Palm to set up analog filters after the oscillators on the follow-up models PPG Wave 2 and PPG Wave 2.2. The result was synthesizers that wrote history and influenced the sound of a whole generation.

In the late 80s, PPG discontinued their work and therefore the production of the Wave, but in the meantime Wolfgang Düren, now manager at Waldorf Electronics, initiated the rebirth of the Wave's technology. Based on an extensive cooperation contract with Wolfgang Palm, the Waldorf MicroWave became the official successor of Wave technology in 1989. The MicroWave was one of the most influental synthesizers of the late 80s and the 90s, right up to today. You can find it on almost any important music production from disco through pop and rock to experimental music. However, the availability of this great synthesizer was not as immediate as was needed, so it was decided in 1995 to further enhance it and to only use those electronic parts that we knew were available. This led to the idea of developing digital filters, and we think we've done a pretty good job.

However, we have not forgotten the past: you can still find the original wavetables of the PPG Wave Computer (Wavetables 001...008), of the PPG Wave 2.2 (009...030, plus the first 8 wavetables) and of the classic MicroWave (031...064, plus 001...030) in the microWAVE II/XT/PC, ensuring that you can still create all famous sounds of those times.

Sound Parameters

Overview of Functions

The Waldorf microWAVE II/XT/PC consists of numerous sound-shaping components. The following overview gives you an idea of how the individual components interact:

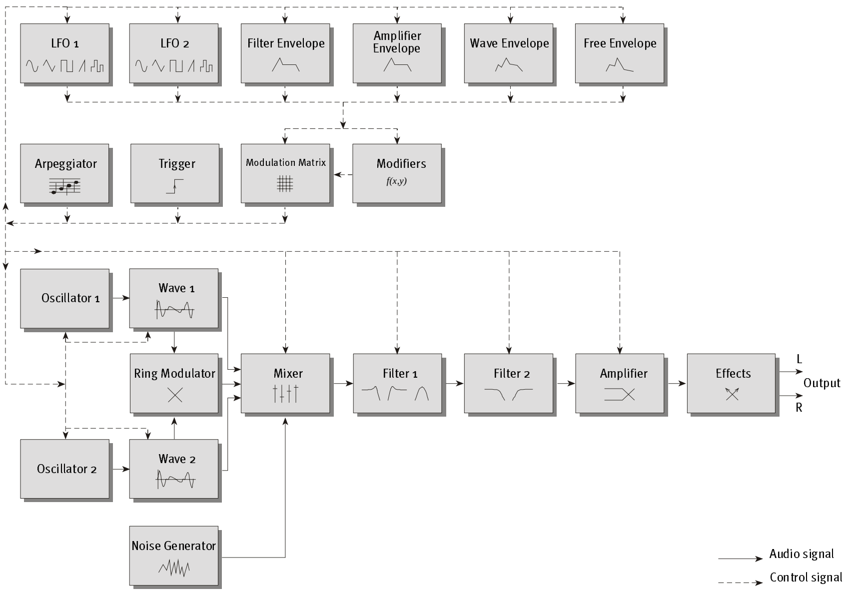


Diagram 1: Block schematic diagram for single sounds

As you can see, the microWAVE II/XT/PC consists of two different types of components:

- Sound generation and sound shaping:
  Oscillators, Waves, Mixer, Filter, Amplifier.
  Sound generation actually occurs within the Waves, which are driven by the Oscillators. They produce a waveform according to the selected wavetable. The Mixer follows the Waves in the signal chain, which is where the Waves' output signals are mixed. Pink noise can also be added to the mix. The Filter then shapes the sound by amplifying (boosting) or attentuating (dampening) certain frequencies. The Amplifier is located at the end of the signal chain, it determines the overall volume and position of the signal within the stereo panorama.
- Modulators: LFOs, Envelopes, Modifiers, Modulation Matrix.
  The Modulators are designed to manipulate or modulate the sound generating components to add dynamics to sounds. The Low-frequency Oscillators (LFOs) are designed for periodic or recurring waveshapes and Envelopes for modulations that occur once within a given time frame. These generators are assigned to parameters via the Modulation Ma-

trix and influence these parameters to alter a sound. In addition, the Modifier unit can process various mathematical operations and functions on the modulation signals.

Oscillators

The oscillators are the first unit in the chain of the microWAVE II/XT/PC's sound generation. In comparison to a classic analog synthesizer, the oscillator's output signal itself is not used as a sound source, it is the driving element for the wavetable synthesis.

Oscillator 1

Osc 1 / 1

Octave 1	Semitone	Detune	Keytrack
-2	+07	+00	+100%

Osc 1 / 2

Pitchbend Range 1	FM Amount
02	010

Octave *-4...+4*

Determines the octave setting of the oscillator. The reference pitch for the oscillator is generated at MIDI note A3 (note no. 69) when **Octave, Semitone** and **Detune** is set to ***0*** and **Keytrack** is ***100%***. In this case the oscillator's frequency will be the same as set in the global **Tune** parameter (normally 440Hz). Set this parameter to ***0*** if you are creating a typical keyboard sound, set it to ***-1*** for bass sounds. If you are programming strings or other high pitched sound, set Octave to ***+1***. The following table shows the relationship between the Octave setting and its corresponding register value, a common measurement based on the length of organ pipes.

Value	Fußlage
-4	128ft.
-3	64ft.
-2	32ft.
-1	16ft.
0	8ft.
+1	4ft.
+2	2ft.
+3	1ft.

Semitone *-12...+12*

Determines the pitch of the oscillator in semitone steps. The standard setting for this parameter is ***0***, but there are cases where different values are required: Most organ sounds include a quint. Therefore one oscillator's semitone parameter must be set to ***+7***. There are also many lead sounds with an interval, e.g. a quart (+5 semitones). When making ring modulated sounds, try to use ***+11*** for the setting.

Detune *-64...+63*

Fine-tunes the oscillator in increments of 128ths of a semitone. The audible result of detuning oscillators is a flanging. Use a positive setting for one oscillator and an equivalent negative setting for the other. A low value of ***±1*** results in a slow and soft flange effect. Mid-ranged settings of ***±5*** are optimal for pads and other fat sounding programs. High values of ***±12*** or above will give a strong detune that can be used for accordeons or effect sounds.

Keytrack *-100% ...+200%*

Determines how much the pitch of the oscillator depends on the MIDI note number. The reference note for Keytrack is E3, note number 64. For positive settings, the oscillator pitch rises on notes above the reference note, for negative settings the oscillator pitch falls up to higher notes and vice versa. A setting of ***+100%*** correspondes to a 1:1 scale, e.g. when an octave is played on the keyboard the pitch changes for the same amount. Other settings than ***+100%*** make sense especially when using ring modulation or oscillator synchronisation. Try to use values in the range ***0...+75%*** or even negative settings for one oscillator while leaving the second at ***+100%*** Keytrack.

Pitchbend Range *0...120 / harmonic / global*

Determines the intensity of the pitchbend via MIDI Pitchbend messages in semitones.

- If ***harmonic*** is selected, the pitchbend is performed in steps of the harmonic and the subharmonic scale. The harmonic scale is used when pitch is bended upwards and built upon multiples of the base pitch. If the base pitch e.g. is 1000Hz, the harmonic scale consists of 2000Hz, 3000Hz, 4000Hz, 5000Hz... and so on. The subharmonic scale is used when pitch is bended downwards and built upon divisions of the base pitch. If the base pitch e.g. is 1000Hz, the subharmonic scale consists of 500Hz, 333.3Hz, 250Hz, 200Hz, 166.7Hz and so on. The following example illustrates the harmonic and the subharmonic scale for the note C3:

  Harmonic scale:C3, C4, G4, C5, E5, G5, A#5, C6, ...

  Subharmonic scale:C3, C2, F1, C1, G#0, F0, ~D0, C0, ...

  Please note that all notes use a pure tuning.
- If ***global*** is selected, the setting in the global parameter **BendRange** is used.

FM Amount *0...127*

Sets the amount that oscillator 2 modulates the frequency of oscillator 1. The sound will get more metallic and sometimes even drift out of tune, especially if oscillator 2 is synced to oscillator 1. To avoid unusable detune, use a triangular or sine like wave for oscillator 2.

Oscillator 2

Osc 2 / 1

Octave 2	Semitone	Detune	Keytrack
+0	+07	+00	+100%

Osc 2 / 2

Pitchbend Range 2	Sync	Link
02	off	on

Octave *-4...+4*
Determines the octave setting of the oscillator. The reference pitch for the oscillator is generated at MIDI note A3 (note no. 69) when **Octave**, **Semitone** and **Detune** is set to 0 and **Keytrack** is ***100%***. In this case the oscillator's frequency will be the same as set in the global **Tune** parameter (normally 440Hz). Set this parameter to ***0*** if you are creating a typical keyboard sound, set it to ***-1*** for bass sounds. If you are programming strings or other high pitched sound, set Octave to ***+1***.

Semitone *-12...+12*
Determines the pitch of the oscillator in semitone steps. The standard setting for this parameter is 0, but there are cases where different values are required: Most organ sounds include a quint, therefore one oscillator's semitone parameter must be set to ***+7***. There are also many lead sounds with an interval, e.g. a quart (+5 semitones). When making ring modulated sounds, try to use ***+11*** for the setting. The semitone setting also becomes very important when oscillator synchronisation is enabled. Then, Oscillator 1 determines the pitch of the generated sound, Oscillator 2 determines the colour. Try to use a random semitone setting while **Octave** is at ***+2***.

Detune *-64...+63*
Fine-tunes the oscillator in increments of 128ths of a semitone. The audible result of detuning oscillators is a flanging. Use a positive setting for one oscillator and an equivalent negative setting for the other. A low value of ***±1*** results in a slow and soft flange effect. Mid-ranged settings of ***±5*** are optimal for pads and other fat sounding programs. High values of ***±12*** or above will give a strong detune that can be used for accordeons or effect sounds.

Keytrack ***-100% ...+200%***

Determines how much the pitch of the oscillator depends on the MIDI note number. The reference note for Keytrack is E3, note number 64. For positive settings, the oscillator pitch rises on notes above the reference note, for negative settings the oscillator pitch falls up to higher notes and vice versa. A setting of ***+100%*** correspondes to a 1:1 scale, e.g. when an octave is played on the keyboard the pitch changes for the same amount. Other settings than ***+100%*** make sense especially when using ring modulation or oscillator synchronisation. Try to use values in the range ***0...+75%*** or even negative settings for one oscillator while leaving the second at ***+100%*** Keytrack.

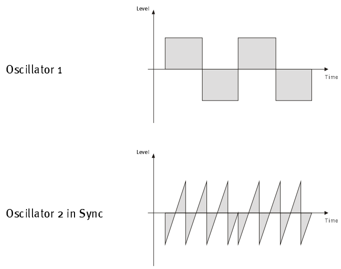
Pitchbend Range ***0...120 / harmonic / global***

Determines the intensity of the pitchbend via MIDI Pitchbend messages in semitones.

- If ***harmonic*** is selected, the pitchbend is performed in steps of the harmonic and the subharmonic scale. Please refer to the description for Oscillator 1 to get further information.
- If ***global*** is selected, the setting in the global parameter ***BendRange*** is used.

Sync ***off / on***

Enables or disables oscillator synchronisation. When enabled, oscillator 2 acts as a slave that is controlled by oscillator 1, the master. Each time oscillator 1 starts a new period, it sends a trigger signal to oscillator 2, forcing it to restart the wave signal, too. As a result, interesting sound effects may be generated, especially when both oscillators are operating at different pitch settings. Using additional pitch modulation by envelopes, LFOs or pitchbend will bring further movement into sync sounds. The following diagram illustrates the principle of oscillator synchronisation in a simplified way:



Link ***off / on***

Allows the same modulation settings for both oscillators to be used. When enabled, oscillator 2 uses the modulation parameters of oscillator 1 for all modulation matrix settings and pitchbend messages. That means, whenever a modulation is applied to oscillator 1, it is also applied to oscillator 2. When disabled, each oscillator uses its own individual modulation settings.

Waves

The waves are the sound sources of the microWAVE II/XT/PC. They are driven by the oscillators' output signal and define the basic spectrum of the generated sound. Please refer to the corresponding topic of this manual to get further information about the wavetable synthesis.

Wave 1

Wave 1 / 1

Startwave	Phase	Wavetable W1
057	132°	001 Resonant

Wave 1 / 2

EnvAmount	EnvVelAmt	Keytrack	Limit W1
20	+15	+068%	off

ⓘ Although the **Wavetable** parameter is the third entry in the Wave 1 / 1 page, it will be explained as the first parameter of these pages. This is because the wavetable defines the basic character of the complete sound. The selected wavetable is used for both wave generators, although it is only displayed in the Wave 1 / 1 page

Wavetable ***001...128***

The Wavetable parameter selects the wavetable for both waves 1 and 2. Each wavetable has a number and a name. The following table shows an overview of all available wavetables and their names:

001	Resonant	017	Formant 1	033	SawSync 1	049	K+Strong2
002	Resonant 2	018	Polated	034	SawSync 2	050	K+Strong3
003	MalletSyn	019	Transient	035	SawSync 3	051	1-2-3-4-5
004	Sqr-Sweep	020	ElectricP	036	PulSync 1	052	19/twenty
005	Bellish	021	Robotic	037	PulSync 2	053	Wavetrip1
006	Pul-Sweep	022	StrongHrm	038	PulSync 3	054	Wavetrip2
007	Saw-Sweep	023	PercOrgan	039	SinSync 1	055	Wavetrip3
008	MellowSaw	024	ClipSweep	040	SinSync 2	056	Wavetrip4
009	Feedback	025	ResoHarms	041	SinSync 3	057	MaleVoice
010	Add Harm	026	2 Echoes	042	PWM Pulse	058	Low Piano
011	Reso 3 HP	027	Formant 2	043	PWM Saw	059	ResoSweep
012	Wind Syn	028	FmntVocal	044	Fuzz Wave	060	Xmas Bell
013	High Harm	029	MicroSync	045	Distorted	061	FM Piano
014	Clipper	030	Micro PWM	046	HeavyFuzz	062	Fat Organ
015	Organ Syn	031	Glassy	047	Fuzz Sync	063	Vibes
016	SquareSaw	032	Square HP	048	K+Strong1	064	Chorus 2

Table 1: Wavetable overview

The wavetables ***065...128*** contain no factory presets. The locations ***065...096*** are reserved for future use. Memory locations ***097...128*** are User Wavetables.

The Wavetables are the real power of the microWAVE II/XT/PC. To make sure that you have access to all this power, you should make yourself familiar with the sound and the characteristic of each wavetable. The best way to do so is to set up a kind of test sound to listen to the wavetables: Start with an initialized sound and turn down the mix level of Oscillator 2. In the Mod Matrix, setup a modulation that uses the ModWheel to modulate ***Wave1Pos*** and set the amount to ***+62*** (the setting of ***+62*** instead of ***+63*** prevents that you accidentally access the "analog" waveforms explained below). Now you can use the Modulation Wheel to sweep through the whole selected wavetable. Change the Wavetable parameter to see how the different wavetables sound. You will notice that they cover an extremely wide range of interesting spectral timbres, including analog, FM-like, bell-type or vocal.

Startwave ***00...60 / triangle / square / sawtooth***

Determines the start point of the wavetable that is used when the sound starts. As an alternative to the waves of the currently selected wavetable, you can select the basic waveforms triangle, ***square*** with 50% duty cycle or ***sawtooth***.

When you want to create a sound with a wave sweep, you should roughly set the Startwave parameter onto the desired wave, before you apply any modulations to the corresponding Wave module. This helps you to find the basic waveform where all modulations start from.

Note that you can apply unipolar and bipolar modulation sources to the Wave module as with any other module. For example, set the Startwave parameter to ***29***, which is almost the middle of the wavetable and apply a slow running LFO to the Wave module to sweep through the whole wavetable (except the three waveforms triangle, square or sawtooth). Try it with one of the PWM wavetables.

ⓘ The basic waveforms triangle, pulse and sawtooth correspond to entry ***61...63*** of each wavetable. Please notice, that these waveforms are also used when an appropriate wave modulation is applied. To avoid this, you will have to activate the **Limit** parameter. Please read this corresponding topic to get further information. Use the basic waveforms to generate traditional, analog synthesizer sounds.

Phase ***free / 3...357°***

By means of this parameter you can define the startsample and, as a result, the phase of the generated wave. Alternative to a fixed value, you can use ***free*** to set the phase to a different, random value each time a note is generated. The setting is scaled in degrees.

EnvAmount ***-64...+63***

Determines the amount of influence the wave envelope has on the wavetable modulation.

EnvVelAmt ***-64...+63***

Determines the amount of influence the wave envelope has on the wavetable modulation, based on key velocity. In conjunction with **EnvAmount** you can create nice effects when you set one of the two parameters to a negative setting while the other one is set to a positive setting.

Keytrack ***-200%...+197%***

Determines the amount of wavetable modulation depending on the received MIDI note number. Reference note for this parameter is E3, note number 64. For positive settings the modulation amount is increased for notes above to reference note, for negative settings the amount is decreased. A setting of +100% corresponds to a 1:1 scale. This means that each note above or below the reference note plays a different wave. E.g., when you set **Startwave** to ***29*** and **Keytrack** to ***+100%***, it means that E3 plays wave 29, F3 plays wave 30, F#3 plays wave 31 and so on.

Limit ***off / on***

This setting prevents, if enabled, accessing the analog type waveforms triangle, square and sawtooth in any case of modulation. When disabled, the full modulation amount will be calculated and applied so that the whole wavetable will be used for tone generation.

Wave 2

Wave 2 / 1

Startwave	Phase	Link	W2
Ø57	free	off	

Wave 2 / 2

EnvAmount	EnvVelAmt	Keytrack	Limit W2
-2Ø	+15	+Ø5Ø%	off

Startwave ***00...60 / triangle / square / sawtooth***

Determines the start point of the wavetable that is used when the sound starts. As an alternative to the waves of the currently selected wavetable, you can select the basic waveforms ***triangle***, ***square*** with 50% duty cycle or ***sawtooth***.

When you want to create a sound with a wave sweep, you should roughly set the Startwave parameter onto the desired wave, before you apply any modulations to the corresponding Wave module. This helps you to find the basic waveform where all modulations start from.

Note that you can apply unipolar and bipolar modulation sources to the Wave module as with any other module. For example, set the Startwave parameter to ***29***, which is almost the middle of the wavetable and apply a slow running LFO to the Wave module to sweep through the whole wavetable (except the three waveforms triangle, square or sawtooth). Try it with one of the PWM wavetables.

ⓘ The basic waveforms triangle, pulse and sawtooth correspond to entry ***61...63*** of each wavetable. Please notice, that these waveforms are also used when an appropriate wave modulation is applied. To avoid this, you will have to activate the **Limit** parameter. Please read this corresponding topic to get further information. Use the basic waveforms to generate traditional, analog synthesizer sounds.

Phase ***free / 3...357°***

By means of this parameter you can define the startsample and, as a result, the phase of the generated wave. Alternative to a fixed value, you can use ***free*** to set the phase to a different, random value each time a note is generated. The setting is scaled in degrees.

Link ***off / on***
Allows the use of the same modulation settings for both waves. When enabled, wave 2 uses the modulation parameters of wave 1 for all Modulation Matrix settings, **EnvAmount**, **EnvVelAmt** and **Keytrack**. That means, whenever a modulation is applied to wave 1, it is also used for wave 2. When disabled, each wave uses its own individual modulation settings.

EnvAmount ***-64...+63***
Determines the amount of influence the wave envelope has on the wavetable modulation.

EnvVelAmt ***-64...+63***
Determines the amount of influence the wave envelope has on the wavetable modulation, based on key velocity. In conjunction with **EnvAmount** you can create nice effects when you set one of the two parameters to a negative setting while the other one is set to a positive setting.

Keytrack ***-200%...+197%***
Determines the amount of wavetable modulation depending on the received MIDI note number. Reference note for this parameter is E3, note number 64. For positive settings the modulation amount is increased for notes above to reference note, for negative settings the amount is decreased. A setting of ***+100%*** corresponds to a 1:1 scale. This means that each note above or below the reference note plays a different wave. E.g., when you set **Startwave** to ***29*** and **Keytrack** to ***+100%***, it means that E3 plays wave 29, F3 plays wave 30, F#3 plays wave 31 and so on.

Limit ***off / on***
This setting prevents, if enabled, accessing the analog type waveforms triangle, square and sawtooth in any case of modulation. When disabled, the full modulation amount will be calculated and applied so that the whole wavetable will be used for tone generation.

QUALITY

The quality parameters control the input stage of the Mixer. They determine the amount of Aliasing and Time Quantization applied to the sound as well as the type of distortion generated when the signal raises the clipping level.

Quality

Aliasing	TimeQuant	Accuracy	Clipping
3	off	off	saturate

Aliasing ***off / 1...5***
Aliasing is a digital side effect that is audible as soon as a wave has harmonics higher than half the sampling frequency. Usually, aliasing is reduced to a minimum by some magical mathematics, but here you can override this and listen to aliasing distortion just like in the dawn of the

first digital musical instruments like the PPG Wave or the first MicroWave. Use a setting other than ***off*** for sounds that expressionally should have a "digital" character

Time Quant ***off / 1...5***

With a wave, 64 harmonics including the fundamental frequency can be represented, and a clever interpolation algorithm makes sure only these 64 harmonics are generated, even at low pitches. However, sometimes one might wish to add additional harshness at the lower end, just like the first MicroWave did, and this is what Time Quantization is for: The wave interpolation is overridden in five steps to get this extra fizziness. Note that pitch accuracy is a bit diminished when using a value other than "off". The audible result of Time Quantization is a very sharp sound character when playing at low pitches. Use this e.g. for sawtooth based sounds..

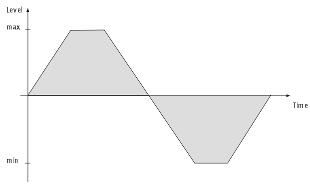
Accuracy ***off / on***

If disabled, voices are detuned very slightly to give more vivid sound, especially when playing chords or sounds with long release. If enabled, the tuning is done as accurate as possible.

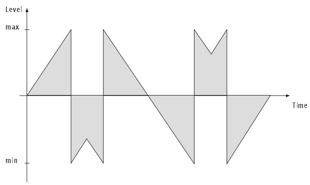
Clipping ***saturate / overflow***

Selects the type of distortion that is applied when the signal raises the clipping level. Clipping is always generated when the sum of all mixer input volumes (i.e. Wave 1, Wave 2 Noise and Ringmodulation) exceeds 128.

- If ***saturate*** is selected for this parameter, the signal will be limited to the maximum level. This is the kind of distortion classic analog circuits will generate.
- If ***overflow*** is selected, distortion is proceeded in the same way as a numerical overflow in a digital system: The polarity of the signal's part above the maximum level will be negated.



saturate



overflow

Mixer

In the mixer you control the volumes of both waves and the noise generator. An optional ring modulation extends the tonal range of the microWAVE II/XT/PC.

Mix

Wave 1	Wave 2	Ringmod	Noise
113	56	Ø	13

Mix 2

External
123

Wave 1 ***0...127***

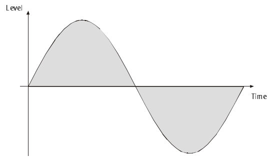
Volume of Wave 1.

Wave 2 ***0...127***

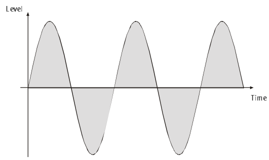
Volume of Wave 2.

Ringmod ***0...127***

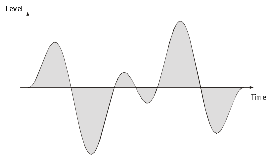
Volume of the ring modulation between Wave 1 and 2. From a technical point of view ring modulation is the multiplication of the waves' signals. The result of this operation is a waveform that contains the sums and the differences of the source frequency components. Since the ring modulation generates disharmonic components, it can be used to add metallic distorted sound characteristics. This is useful e.g. when generating synth percussion. The following diagram illustrates what happens when two sine waves are ring modulated. Please note that in a complex waveform all harmonic component behave like interacting sine waves, resulting in a wide spectral range of the ring modulated sound.



Wave 1
$f_1 = 1$



Wave 2
$f_2 = 2{,}5$ (1 octave + 4 semitones)



Ring Modulation
$f_{res1} = f_1 - f_2 = 1{,}5$
$f_{res2} = f_1 + f_2 = 3{,}5$

Noise ***0...127***

Volume of the noise generator. The noise generator produces pink noise and features no other controls. Noise is a fundamental source for any kind of analog-type percussion. Also wind and other sound effects can be created by using the noise generator.

External ***0...127***

Volume of the external audio **input** (digital input on microWAVE PC).

Play Access

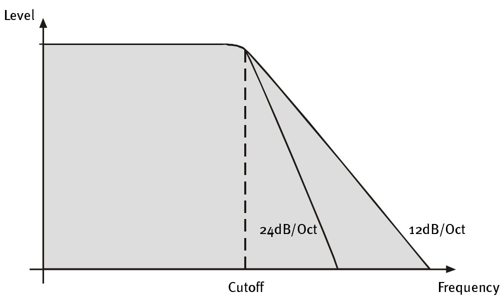
The Play Access page is a very exciting feature that gives you an easy accessible control over 4 freely-definable Sound parameters. This can be extremly useful in adapting a sound very quickly as well as having easy realtime control in performance situations.

There is no need to use the Play Access mode with the microWAVE PC.

Filter

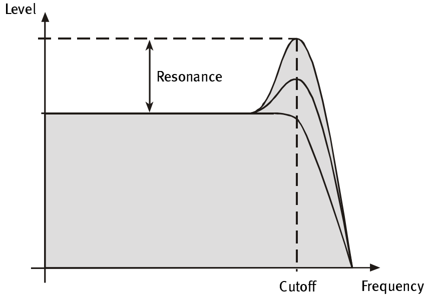
Once the audio signal leaves the mixer, it is sent to the filters. The microWAVE II/XT/PC has two independent filter units, each with its own individual settings. Both filters are routed in series. The filters are components that have significant influence on the microWAVE II/XT/PC's sound characteristics.

The filter type most commonly used in synthesizers is a low pass filter. This type dampens frequencies that lie above a specified cutoff frequency. Frequencies below this threshold are hardly affected. The frequency below the cutoff point is called the pass band range, the frequencies above are called the stop band range. The microWAVE II/XT/PC's filter dampens frequencies in the stop band with a certain slope. The slope is selectable between 12dB and 24dB per octave. This means that the level of a frequency that lies an octave above the cutoff point will be 12dB or 24dB less than those frequencies of the signal that fall into the pass band. The following diagram shows the basic principle of a low pass filter:



To give you an idea of the extent of damping, consider this: A reduction of 24dB reduces the original level by approx. 94%. The damping factor two octaves above the cutoff point reduces the original level by more than 99%, which in most cases means this portion of the signal is no longer audible.

The microWAVE II/XT/PC's filter also features a resonance parameter. Resonance in this context means that a narrow frequency band around the cutoff point is emphasised. The following diagram shows the effect of the resonance parameter on the filter's frequency curve:



If the resonance is raised to a great extent, then the filter will begin self-oscillation, i.e. the filter generates an audible sine wave even when it does not receive an incoming signal.

Filter 1

Filter 1 gives you the most flexibility by offering low pass, high pass and band pass types. In addition, there is an sine waveshaping filter with an 12dB low pass following. You can select the slope between 12dB and 24dB per octave for the low pass and band pass. Further types might be added in the futur.

Filter 1 / 1

Cutoff	Resonance	Type	Keytrack
047	012	24dB LP	+066%

Filter 1 / 2

Cutoff Env. Amount	Env.Velocity Amount
69	-23

Cutoff ***0...127***

Determines the cutoff frequency for the low pass and high pass filter types and the mid frequency for the band pass type. When a low pass is selected via the **Type** parameter, all frequencies above the cutoff frequency are damped. When high pass is selected, all frequencies below the cutoff frequency are damped. In a band pass only frequencies near the cutoff setting will be passed through. You can bring more movement into the sound by modulating the cutoff frequency via the LFOs, the envelopes or the Keytrack parameter. At a value of ***64*** and a **Resonance** value of ***114***, the filter oscillates with 440Hz, which is equal to A3. Tuning is scaled in semitone steps. When **Keytrack** is set to ***+100%*** , the filter can be played in a tempered scale.

Resonance ***0...127***

Filter resonance parameter. Determines the amplification of the frequencies around the cutoff point. Use lower values in the range ***0...80*** to give more brilliance to the sound. At higher values of ***80...113*** the sound gets the typical filter character with a strong boost around the cutoff frequency and a loss in the other range. When the setting is raised to values above ***113***, the filter starts to self-oscillate, generating a pure sine wave. This feature can be used to create solo sounds like the traditional "moog lead" or analog-style effects and percussion like electronic toms, kicks, zaps etc.

Type ***siehe Tabelle***

Selects the filter type. Further information on the different filter types is given at the end of this chapter.

Keytrack ***-200%...+197%***

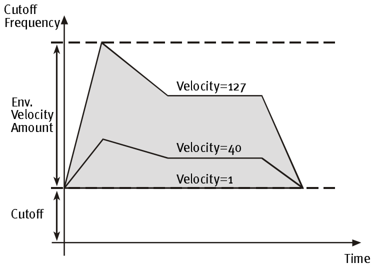
Determines how much the cutoff frequency depends on the MIDI note number. The reference note for Keytrack is E3, note number 64. For positive settings, the cutoff frequency rises on notes above the reference note, for negative settings the cutoff frequency falls up to higher notes and vice versa. A setting of ***+100%*** corresponds to a 1:1 scale, so e.g. when an octave is played on the keyboard the cutoff frequency changes for the same amount. If you want to play the filter in a tempered scale, e.g. for a solo sound with self-oscillation, set the value to ***+100%***. On most bass sounds lower settings in the range ***+60...+75%*** are optimal to keep the sound smooth at higher notes.

Cutoff Env. Amount ***-64...+63***

Determines the amount of influence the filter envelope has on the cutoff frequency. For positive settings, the filter cutoff frequency is increased by the modulation of the envelope, for negative settings, the cutoff frequency is decreased. Use this parameter to change the timbre of the sound over time. Sounds with a hard attack usually have a positive envelope amount that makes the start phase bright and then closes the filter to get a darker sustain phase. On the other side string sounds usually use a negative envelope amount that gives a slow and dark attack before the cutoff rises in the sustain phase.

Env. Velocity Amount *-64...+63*

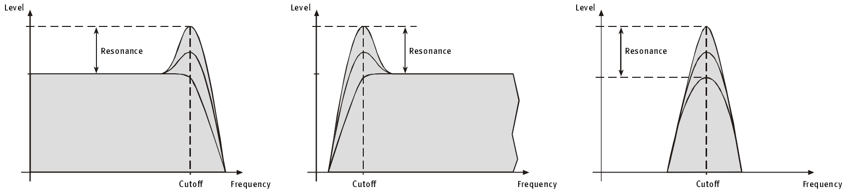
Determines the amount of influence the filter envelope has on the cutoff frequency, based on key velocity. This parameter works similarly to the **Cutoff Env. Amount** parameter with the difference that its strengh is velocity based. Use this feature to give a more expressive character to the sound. When you hit the keys smoothly, only few modulation is applied. When you hit them harder, the modulation amount also gets stronger. The following diagram illustrates the functionality of this parameter:



ⓘ The overall modulation applied to the filter's cutoff frequency is calculated as the sum of both parameters **Cutoff Env. Amount** and **Env. Velocity Amount**. Therefore you should always bear in mind what the result is, especially when the filter does not behave as you expect. You can also create interesting effects by setting one parameter to a positive amount and the other to a negative.

Filter Types

This paragraph describes the microWAVE II/XT/PC's different filter types. Most types are based on traditional low pass, high pass or band pass structures. The following diagram illustrates the frequency plots of these types:



Low pass High pass Band pass

The filter types have the following display designations:

Setting	Filter Type
24dB LP	24dB low pass
12dB LP	12dB low pass
24dB BP	24dB band pass
12dB BP	12dB band pass
12dB HP	12dB high pass
Sin(x)>LP	Sinus-Waveshaper followed by a 12dB low pass
WaveShapr	12dB low pass filter with wave shaper
Dual L/BP	Parallel 12dB lowpass/bandpass filters
FM-Filter	12dB low pass filter with frequency modulation
S&H>L12dB	Sample-and-hold in front of 12dB low pass filter

When some of the above types are selected, an extra parameter appears on the ***Filter 1*** / 2 page. Exactly what this parameter is for depends on the type of filter selected. The extra parameter is therefore described together with every new filter type

Modulation of the "Extra" Parameter

The "extra" parameter of the filter types described below may be selected in the modulation matrix and is designated as ***F1 Extra***. (An abbreviation of "Filter 1 Extra Parameter")

ⓘ Do not mistake "FM Amount" for filter ***FM amount***. The filter FM amount is the ***F1 Extra*** modulation destination whenever the FM-filter is selected on the ***Filter 1 / 1*** page. The ***FM Amount*** destination in the modulation matrix is for oscillator FM.

224db Low Pass and 12dB Low Pass

The low pass types ***24dB LP*** and ***12dB LP*** are suitable for the most usual applications. Use the 24dB slope if you want to create sounds with a typical audible filtered character, use the 12dB slope if you want to get softer results.

24db Band Pass and 12dB Band Pass

The band pass filters ***24dB BP*** and ***12dB BP*** remove frequencies both below and above the cutoff point. As a result, the sound character gets narrow. Use these filter types for programming effect and percussion-like sounds.

12db High Pass

The high pass filter ***12dB HP*** is useful to thin out a sound's bass frequencies. This may give interesting results also in conjunction with cutoff frequency modulation. By doing this you can e.g. "fly-in" a sound starting at its high harmonics and then coming up to its full frequency range.

Sine Waveshaper with 12dB Low Pass

The ***Sin(x)>LP*** Type consists of a sine waveshaper followed by a 12dB low pass filter with resonance. The sine waveshaper usually adds some harmonics and intermodulation frequencies to the signal.

12dB Low-pass Wave Shaper

This new filter type consists of two components, the first being a normal 12dB low-pass filter as described in the user manual. The second component is a wave-shaper much like the sine waveshaping filter ***Sin(x)>LP*** also described in the manual. The difference between the sine wave shaper and this new shaper is that the shaping wave is no longer a sine wave but a wave from the wavetable used by the sound.

The extra parameter ***Wave***, on the ***Filter 1 / 2*** page is used to select the desired shaping wave from the sound's wavetable (e.g. a triangle wave):

Filter 1 / 2

Cutoff Env. Amount	Env.Velo	Wave
69	-23	triangle

ⓘ For a nice gritty sound, try a square wave as shaping wave!

12 dB parallel Low-pass and Band-pass Filters

This filter type consist of two filters parallel to each other. The first filter being of the low-pass type and the second of the band-pass type. As with the new wave shaping filter, the 12 dB low-pass filter can be adjusted the usual way as described in the user manual.

The band-pass filter's cutoff frequency is the same as the cutoff frequency of the low-pass filter cutoff setting except for the extra parameter ***BP Offset***, which adds to the band-pass filter's cut-off frequency. The band-pass filter's resonance is equal to that of the low-pass filter.

Filter 1 / 2

Cutoff Env. Amount	Env.Velo	BP Offset
69	-23	+14

To select a low-pass/band-pass with the latter set to one octave above the other, do the following:

1. Go to the ***Filter 1 / 1*** page and select the ***Dual L/BP*** filter type
2. Then go to the ***Filter 1 / 2*** page. The third parameter should now read ***BP Offset***. Change this setting so that it reads ***+12***.

Because the BP offset is in semitones, the band-pass filter's cutoff frequency is now an octave above the low-pass filter's cutoff frequency.

12 dB Low-pass Filter with Frequency Modulation

The FM-filter type is a 12dB low-pass filter where the cutoff frequency can be modulated by the output of oscillator 2. The filter may be setup exactly like a normal low-pass filter.

The modulation amount ***Osc2 FM*** is the extra parameter and can be found on the ***Filter 1 / 2*** page:

Filter 1 / 2

Cutoff Env. Amount	Env.Velo	Osc2 FM
69	-23	Ø78

Sample-and-hold 12dB Low-pass Filter

The S&H-filter has a sample-and-hold (S&H) circuit with adjustable rate in front of the 12 dB low-pass filter. The S&H circuit effectively lowers the sampling rate so that the harmonics are reflected to another frequency producing a harsh sound.

The rate of the S&H circuit is the extra parameter and appears on the ***Filter 1 / 2*** page as ***S&H Rate***. When the ***S&H rate*** is set to maximum (127), the circuit passes the sound untouched.

Filter 1 / 2

Cutoff Env. Amount	Env.Velo	S&H Rate
69	-23	Ø69

ⓘ If you like nice clean sounds, the S&H filters are definitely not for you.

Filter 2

The second filter is capable of performing a low pass or high pass. The slope is always 6dB per octave, there is no resonance parameter and therefore no self-oscillation. You can use Filter 2 in several ways. Since its slope is more flat than those of Filter 1, the effect filtering has on the sound is very subtle.

Filter 2

Cutoff Filter 2	Type	Keytrack
102	6db LP	+000%

Cutoff *0...127*

Determines the cutoff frequency. Note that you can also modulate the filter's cutoff frequency in the modulation matrix.

Type ***6dB LP / 6dB HP***

Selects the filter type.

- Use the low pass setting ***6dB LP*** to get a warm sound without cutting of too much of the higher frequencies.
- Use the high pass setting ***6dB HP*** to thin out the bass frequencies in order to get a cleaner and more precious sound.

Keytrack ***-200%...+197%***

Determines how much the cutoff frequency depends on the MIDI note number. The reference note for Keytrack is E3, note number 64. For positive settings, the cutoff frequency rises on notes above the reference note, for negative settings the cutoff frequency falls up to higher notes and vice versa. A setting of ***+100%*** corresponds to a 1:1 scale, so e.g. when an octave is played on the keyboard the cutoff frequency changes for the same amount.

ⓘ WIf you don't want to use Filter 2, select the low pass and set the cutoff frequency to ***127***.

Volume and Pan

This unit is the last part in the microWAVE II/XT/PC's internal signal routing. Its purpose is to set the volume and the pan position of the sound. After that the signal passes the D/A converter and can be taken from the audio jacks on the rear panel (in microWAVE PC it goes straight to the EWS's digital input on the card, or to the digital outs on your front module).

To understand the operation of this unit, it is important to know that the Amplifier Envelope is always acting as a modulation source for the volume. This means that an audio signal can only pass through if the Amplifier Envelope is triggered and opened.

Finally a chorus or a ensemble effect can be added to enhance the sound.

Volume

Amplifier

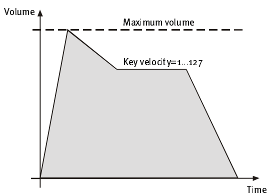
Volume	Velocity	Keytrack	Effect
090	+48	+000%	Chorus

Volume ***0...127***

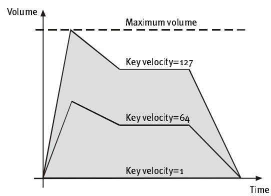
Determines the master volume of the sound program.

Velocity ***-64...+63***

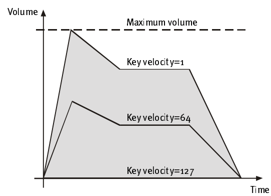
Specifies how much volume will be affected by keyboard velocity. Use this feature to give more expression to the sound. With a setting of 0, velocity will have no effect on the volume. Classic organs work in this way because they do not have dynamic response. For positive settings, the volume rises up to higher velocities. This is the most commonly used setting which gives a piano-like character. For negative settings, the volume falls up to higher velocities. This gives an untypical character suitable for effect sounds. As the Amplifier always works in conjunction with the Amplifier Envelope, this parameter actually determines the envelope velocity amount. The following diagram illustrates this functionality:



Velocity parameter = 0



Velocity parameter = +63



Velocity parameter = -64

Keytrack ***-200% ...+197%***
Determines how much the volume depends on the MIDI note number. The reference note for Keytrack is E3, note number 64. For positive settings, the volume increases on notes above the reference note, for negative settings the volume decreases up to higher notes and vice versa. This setting can be useful to adjust a sound's volume over the whole keyboard range. Especially when extensive filtering is used, the sound can be louder on the lower or the upper part of the keyboard. On the other side, you can apply this effect intentionally e.g. for effect sounds.

Effect ***off / Chorus / Ensemble***
Enables and selects the type of effect that is used for the sound program. You can choose between a chorus and an ensemble effect.

- The chorus consists of two short delays where delay time is modulated with a sine wave of about 0.5 Hz. It spreads the stereo image of the program by giving it a wide sounding character.
- Ensemble is similar except it has more delays and higher modulation frequencies. This effect is useful in combination with strings or other pad sounds.

Pan

Pan

Panning	Keytrack
left 50	+200

Panning ***left 64...center...right 63***
Determines the position in the stereo panorama. When the setting is ***left 64***, the sound is panned far left, when the setting is ***right 63***, it is panned far right. If you want to set the sound into the middle of the stereo panorama, use the ***center*** setting. To give further movement to the sound, set this parameter to a basic value and apply some modulation to it e.g. via an LFO or the **Keytrack** parameter.

Keytrack ***-200% ...+197%***
Determines how much the pan position depends on the MIDI note number. The reference note for Keytrack is E3, note number 64. For positive settings, the panning moves to the right on notes above the reference note, for negative settings the panning moves to the left up to higher notes and vice versa. This feature enables you to give a typical piano-like panning, where lower notes are on the left side and higher notes on the right. To achieve this, set the **Panning** parameter to ***center*** and Keytrack to ***+197***.

Effects

All the effect parameters are available on the ***Effect*** page which is located between the ***Amplifier*** and ***Pan*** page. The first parameter on the Effect page is always the effect type parameter. The other three parameters change according to the type of effect which has been selected.

Some Words about Effects

It is very difficult to describe effects such as chorus and flanger. Therefore, the description of the exact timbre changes induced by the effects has been omitted. As it would serve no purpose to clutter the manual with subjective obscurity. Just have a play with the effects!

The Mix Parameter

Most of the effects have a ***mix*** parameter. This parameter determines the volume ratio between the original signal and the effect output. To further stress the fact that this is a ratio, the mix parameter is display as two numbers. The first number is the original or dry signal amount. The second number is the effect's output amount, or wet signal amount. The two numbers are separated by a colon (see chorus display example).

Chorus

Below, the display of the Microwave is shown with the Chorus effect selected:

Effect

Effect	Speed	Depth	Mix
Chorus	052	048	0:127

Speed ***0...127***
Determines the oscillator speed of the chorus effect.

Depth ***0...127***
Determines the amount of the chorus.

Mix ***127:0...0:127***
Determines the volume ratio of the dry and wet signal.

Flanger 1

Effect

Effect	Speed	Depth	Mix
Flanger 1	052	048	0:127

Speed ***0...127***
Determines the oscillator speed of the flanger effect.

Depth ***0...127***
Determines the amount of flanging.

Mix *127:0...0:127*
Determines the volume ratio of the dry and wet signal.

Flanger 2

Effect

Effect	Speed	Feedback	Mix
Flanger 1	Ø38	1ØØ	55:72

Speed *0...127*
Determines the oscillator speed of the flanger effects.

Feedback *0...127*
Determines the amount of feedback.

Mix *127:0...0:127*
Determines the volume ratio of the dry and wet signa.

AutoWahLP

Effect

Effect	Sense	Cutoff	Resonance
AutoWahLP	Ø65	Ø38	Ø1Ø

The AutoWahLP is basically a low-pass filter of which the cutoff is determined by the signal's strength.

Sense *0...127*
Controls the filter's sensitivity according to the signal's strength.

Cutoff *0...127*
The minimal cutoff frequency of the filter.

Resonance *0...127*
Filter resonance.

AutoWahBP

Effect

Effect	Sense	Cutoff	Resonance
AutowahBP	Ø65	Ø38	Ø1Ø

The AutoWahBP is basically a band-pass filter of which the cutoff is determined by the signal's strength.

Sense *0...127*
Controls the filter's sensitivity according to the signal's strength.

Cutoff *0...127*
The minimal cutoff frequency of the filter.

Resonance *0...127*
Filter resonance.

Overdrive

Effect

Effect	Drive	Gain	Amp Type
Overdrive	Ø18	Ø93	Combo

Drive *0...127*
Determines how much distortion is applied.

Gain *0...127*
Determines the output volume of the distortion.

Amp Type *0...127*
Allows one to select the speaker simulation setting. These settings are available:

Setting	Type of Simulation
Direct	No speaker simulation
Combo	Simulation of a small speaker with small bandwidth
Medium	Simulation of a larger speaker with medium bandwidth
Stack	Simulation of an array of speakers with large bandwidth

EAmp. Mod

Effect

Effect	Speed	Spread	Mix
Amp. Mod	Ø38	1ØØ	55:72

The Amplitude Modulator can be used as a tremolo or as a low-frequency ring modulator. For use as a tremolo, the dry signal (the first number of the Mix parameter) must be kept above ***63***. For use as a ring modulator, the dry signal must be kept below ***64***.

Speed *0...127*
Oscillator speed of the amplitude modulator.

Spread *0...127*
Amount of lag between the left and right channel.

Mix *127:0...0:127*
Determines the volume ratio of the dry and wet signal.

Delay

Effect

Effect	Time	Feedback	Mix
Delay	1/4 [74]	090	106:21

Time

Delay time. This parameter is displayed as a note type followed by a Beats-Per-Minute number. So ***1/4 [74]*** means that the delay time is a quarter-note at 74 BPM.

Feedback ***0...127***

Determines the amount of delayed signal being fed back into the delay.

Mix ***127:0...0:127***

Determines the volume ratio of the dry and wet signal.

Pan Delay

Effect

Effect	Time	Feedback	Mix
Pan Delay	1/4 [74]	090	106:21

The only difference between Delay and Pan Delay is that the delayed signal seems to bounce from the left channel to the right and back again.

Mod Delay

Effect

Effect	Time	Speed	Depth
Mod Delay	1/4 [74]	010	108

The modulated delay is a delay type effect where the delay time is modulated by a low frequency oscillator. The speed of the oscillator and the amount of change caused by the oscillator are parameters of this effect.

Time

Delay time. This parameter is displayed as a note type followed by a Beats-Per-Minute number. So ***1/4 [74]*** means that the delay time is a quarter-note at 74 BPM.

Speed ***0...127***

The speed of the modulating oscillator.

Depth ***0...127***

Amount of change in the delay time caused by the oscillator.

Portamento and Glissando

The term "portamento" describes the continuous gliding from one note to the next like strings or some brass instruments (e.g. trombone) can do. A glissando is a similar effect with one difference: The pitch does not change continuously but in note steps. On acoustic instruments a glissando can be performed e.g. on a piano when you play very fast over a wide key range. The microWAVE II/XT/PC offers some different effect types that can be trimmed for each situation. The term "glide" is used for all different types of effect in common.

Glide

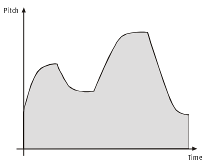
Active	Type	Mode	Time
on	Gliss	exp,	25

Active ***off / on***
Enables or disables the glide effect.

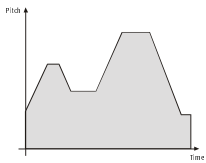
Type ***porta / glissando / fingered / f.gliss***
Determines the effect type.

- ***Porta*** selects a normal portamento effect with all notes gliding continously from one to the next.
- Similar to that, ***gliss*** selects the normal glissando effect with all notes gliding in semitone steps.
- When ***fingered*** or ***f.gliss*** is selected, the portamento or glissando is only applied on legato played notes and so the first note played is not influenced. This feature is useful especially for solo sounds, when it is often undesireable to slide into the beginning.

Mode ***exp. / linear***
Selects whether the pitch is changed in an exponential or linear style. On classic analog synthesizer the ***exponential*** style was used mainly since it could be easily created with analog circuits. The ***linear*** setting produces a more accurate gliding with better audible results. The following diagram illustrates the difference between the two modes:



exponential



linear

Time ***0...127***
Determines the glide time. Low values will give a short glide time in the range of milliseconds that gives a special character to the sound. High values will result in a long glide time up to several seconds which can be useful for solo and effect sounds.

Trigger

The Trigger parameters define how the various envelopes are started. In addition, you can activate special dual and unisono modes to stack the microWAVE II/XT/PC's voices.

Trigger 1

FilterEnv	Amp. Env	Wave Env	Free Env
normal	single	normal	retrigger

Trigger 2

Mode	Assign	Detune	De-Pan
Poly	unisono	Ø25	11Ø

FilterEnv ***normal / single / retrigger***

Determines the way of triggering the Filter Envelope.

- If ***normal*** is selected, every note triggers the envelope of its own voice.
- If ***single*** is selected, the envelopes of all voices act as one. The envelope is started, when the first note is played. The sustain phase is held until the last note is released. Then the release phase is performed.
- If ***retrigger*** is selected, the envelope acts as in single mode except that each note triggers the envelope again from its current value.

Amp. Env ***normal / single / retrigger***

Determines the way of triggering the Amplifier Envelope.

- If ***normal*** is selected, every note triggers the envelope of its own voice.
- If ***single*** is selected, the envelopes of all voices act as one. The envelope is started, when the first note is played. The sustain phase is held until the last note is released. Then the release phase is performed. This setting is only valid, if **Mode** is set to ***Mono***. Otherwise the envelope works as if ***normal*** is selected.
- If ***retrigger*** is selected, the envelope acts as in ***single*** mode except that each note triggers the envelope again from its current value. This setting is only valid, if **Mode** is set to ***Mono***. Otherwise the envelope works as if ***normal*** is selected.

Wave Env ***normal / single / retrigger***

Determines the way of triggering the Wave Envelope.

- If ***normal*** is selected, every note triggers the envelope of its own voice.
- If ***single*** is selected, the envelopes of all voices act as one. The envelope is started, when the first note is played. The sustain phase is held until the last note is released. Then the Key-off phase is performed.
- If ***retrigger*** is selected, the envelope acts as in ***single*** mode except that each note triggers the envelope again from its current value.

Free Env. ***normal / single / retrigger***

Determines the way of triggering the Free Envelope.

- If ***normal*** is selected, every note triggers the envelope of its own voice.
- If ***single*** is selected, the envelopes of all voices act as one. The envelope is started, when the first note is played. The sustain phase is held until the last note is released. Then the release phase is performed.
- If ***retrigger*** is selected, the envelope acts as in ***single*** mode except that each note triggers the envelope again from its current value.

Mode ***Poly / Mono***

Selects whether the sound can be played polyphonic or monophonic.

- Use the ***Poly*** setting for normal applications when you where to play chords.
- If ***Mono*** is selected, the microWAVE II/XT/PC playes only the last incoming note. Use this mode for solo sounds, especially in combination with the Glide effect.

Assign ***normal / dual / unisono***

Defines who the sound's voices are assigned to the played notes.

- If ***normal*** is selected, every played note uses one of the microWAVE II/XT/PC's voices.
- If ***dual*** is selected, every note uses two voices which can be detuned by the ***Detune*** parameter described below.
- If ***unisono*** is selected, all voices are used, divided to the notes played. That means, if you play just one note, all 10 voices of the microWAVE II/XT/PC are used for this note. If you play two notes, 5 voices are used for each note and so on. The ***Detune*** parameter is also active in this mode.

Detune ***0...127***

Determines the amount of oscillator detune when dual or unisono is selected in the Assign parameter. The setting always represents the maximum detune range of all used voices. E.g. in dual mode a value of ***40*** means a detune of ***-20*** for the first voice and ***+20*** for the second.

De-Pan ***0...127***

If dual or unisono is selected, the voices are spread in panorama according to this parameter. Use 127 to get a full spread or 0 to get no spread at all. If neither dual nor unisono is selected, the setting of this parameter has no audible effect.

Arpeggiator

An arpeggiator is a device that splits an incoming MIDI chord into its single notes and repeats them rhythmically. Different sequence modes can be defined for the arpeggiator to cover a wide range of applications.

In addition to the synthesis features, the microWAVE II/XT/PC offers a separately programmable arpeggiator for every sound program. The arpeggiator can be used independently or synced to MIDI clock. It can play a wide range of different rhythm patterns, including a user programmable.

The arpeggiator uses an internal buffer that can store up to 20 notes. The buffer is cleared each time a new chord is played. There are two ways of entering a chord:

- Press all keys of the chord simultaniously.
- Press and hold the first key of the chord. While holding this key, enter the other keys sequentially. After playing all keys, release the first key again. On one hand this method is practicable for playing difficult chords, on the other hand it is essential when using the ***as played*** setting of the **Direction** parameter. This setting allows you to create arpeggios in the sequence of played notes.

When you use the sound as part of a multi program, you can either use the sound's arpeggiator described here, or the dedicated arpeggiator of the multi program's instrument. Use the instrument parameter **Arpeggiator Active** to select which one to use. As a default the sound's arpeggiator is not activated and therefore no arpeggio will be generated when turning on the arpeggiator here.

Arpeggiator 1

Active	Tempo	Clock	Range
on	126	1/16	04

Arpeggiator 2

Pattern	Direction	NoteOrder	Velocity
on	alternate	as played	last note

Arpeggiator 3

Reset on Pattern Start	Length
off	08

Arpeggiator User Pattern

Position	Trigger
03	on [*-*--**-]

Active ***off / on / hold***

Enables or disables the arpeggiator or activates the hold mode. When ***hold*** is activated, incoming MIDI chords generate continuous arpeggios even when the chord is released. The microWAVE II/XT/PC will continue to do so until you play a new chord or this parameter is set back to ***off*** or ***on***. You can also stop the arpeggiator by performing the panic function or sending an All Notes Off message from your sequencer.

Tempo ***extern / 50...300***

Sets the arpeggiator's basic tempo. Can be defined manually in BPM (beats per minute) or via MIDI clock, if ***extern*** is selected.

ⓘ The arpeggiator can be used as a master as well as a slave via the MIDI clock:

- When you use the arpeggiator as the master, set its speed via the Tempo parameter. Set the global parameter **MIDI Clock Send** to ***on***. This enables the sending of MIDI clock signal via the microWAVE II/XT/PC's MIDI out jack H.
- When you use the arpeggiator as a slave, an external device (e.g.sequencer) determines the tempo of the arpeggiator. Set the Tempo parameter to ***external*** as described above. Here, too, notes and MIDI clock information can be used to control other devices. In this mode, the MIDI Song Position Pointer is also recognized.

Clock ***1/1...1/32***

Determines the note value for whole notes to thirty-second notes. The basis is a 4/4 beat. Triplets (e.g. ***1/8T***) and dotted notes (e.g. ***1/16.***) are available for every value.

Range ***1...10***

Determines the range of the single notes in octaves.

Pattern ***off / user / 1...15***

Determines whether an rhythm pattern is played and which one.

- If ***off*** is selected, the arpeggiator playes its notes in regular steps, specified by the **Clock** parameter.
- If ***user*** is selected, the arpeggiator uses the free programmable pattern defined in the ***Arpeggiator User Pattern*** page.
- Additionally, the arpeggiator features 15 preset rhythm patterns. These are numbered from ***1*** through ***15***. Here is an overview of the arpeggiator preset patterns:

Pattern	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
1	●		●	●	●		●	●	●		●	●	●		●	●
2	●		●		●			●	●		●		●			●
3	●		●		●		●	●	●		●		●		●	●
4	●		●	●	●		●		●		●	●	●		●	
5	●		●		●	●		●	●		●		●	●		●
6	●	●		●		●	●		●	●		●		●	●	
7	●		●		●		●		●	●		●		●		●
8	●		●		●		●	●		●		●	●		●	
9	●	●	●		●	●	●		●	●	●		●	●	●	
10	●	●		●	●		●	●		●	●		●	●	●	
11	●	●		●	●		●	●		●	●		●		●	
12	●	●		●	●		●		●	●		●	●		●	
13	●		●		●		●		●	●		●		●	●	●
14	●			●			●			●			●			●
15	●		●		●		●		●			●	●		●	

Diagram 2: Arpeggiator patterns

Direction *up / down / alternate / random*

Determines the sequence of generated notes according to pitch.

- If ***up*** is selected, the arpeggio starts at the lowest note and sweeps up through the notes until it reaches the highest note. It then starts at the bottom again.
- If ***down*** is selected, the arpeggio starts at the highest note and sweeps down through the notes until it reaches the lowest note. It then starts at the top again.
- If ***alternate*** is selected, the arpeggio starts at the lowest note and sweeps up through the notes until it reaches the highest note. It then starts to sweep back down.
- If ***random*** is selected, the arpeggio plays any of the notes in a random order.

NoteOrder *by note / note rev. / as played / reversed*

Determines the sequence of generated notes according to note order.

- If ***by note*** is selected, the arpeggio sequence is sorted by the MIDI note number. This is the standard mode, used by most arpeggiators.
- If ***note rev.*** is selected, the arpeggio sequence is sorted in the exactly reversed order to the ***by note*** setting.
- If ***as played*** is selected, the arpeggio is generated in the order of the incoming notes. In combination with the user programmable pattern this feature offers a small but effective step sequencer.
- If ***reversed*** is selected, the arpeggio is generated in the reverse order of the incoming notes.

To understand the difference of the individual settings, it is nessessary to "step-input" the notes of the chord as described at the beginning of this chapter.

Velocity *root note / last note*

Determines how the velocity values of the generated notes are calculated.

- If ***root note*** is selected, every generated note inherits its velocity from its base note. E.g. if the base chord for the arpeggio contains an E with a certain velocity, all generated E notes also have this velocity value, independent of their octave setting.
- If ***last note*** is selected, every generated note has the same velocity as the last incoming note.

Reset on Pattern Start *off / on*

Selects if the arpeggiator is reset each time the rhythm pattern starts again. If the setting is disabled, the arpeggiator plays all chord notes from the first to the last and over again, regarding the sequence determined by **Direction** and **Note Order**. If the setting is enabled, the arpeggiator only plays the number of chord notes that correspond to the pattern length. Then it starts with the first chord note at its basic octave again. The result is similar to pressing the chord again each time the pattern restarts. If no pattern is selected, this parameter has no function.

Length *1...16*

Determines the length of the user programmable rhythm pattern.

Position *1...pattern length*

Trigger ***off / on***

These two parameters are used to define the user programmable rhythm pattern. Before entering the pattern, you must set its length via the **Length** parameter. Use the **Position** parameter to select the position of the pattern you want to edit. Then use the **Trigger** parameter to define the state of the selected position. All active positions are marked with a "*" in the display, all inactive positions show a "-". Note that you can also create triplet rhythms by setting the pattern length to 3, 6 or 12 and selecting a triplet value for the **Clock** parameter.

Arpeggiator User Pattern

Position	Trigger
Ø3	on [*-*--**-]

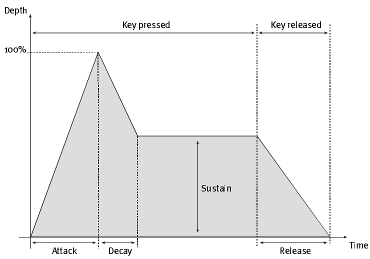
ⓘ In the microEdit software you can click directly into the display to program the arpeggiator steps.

ENVELOPES

The microWAVE II/XT/PC's envelopes allow you to manipulate sound parameters via rate or timed modulations. The microWAVE II/XT/PC offers four independent programmable envelopes for every sound program:

- A filter envelope with ADSR characteristic
- A volume envelope with ADSR characteristic
- A wave envelope with 8 different times and levels (multi segment envelope)
- An additional free multi segment envelope with 3 different times and levels and a release time and release level

ⓘ Most traditional synthesizers feature ADSR envelopes. These envelopes are made up of four parameters that determine their response: **Attack, Decay, Sustain** and **Release**. The following diagram illustrates the structure of an ADSR envelope:



The envelope is started by pressing a key. It ascends to its maximum value at the rate determined by the **Attack** parameter. It then descends at the rate determined by the **Decay** value until it reaches the predetermined **Sustain** value. It remains at this value until the key is released. The envelope then descends to zero at the rate determined by the **Release** parameter.

Filter Envelope

This envelope is designed to control the filter but can also be used for other modulations. The following parameters determine the envelope's response:

Filter Env

FE Attack	Decay	Sustain	Release
000	035	090	020

Attack *0...127*

Determines the attack rate or amount of time it takes for a signal to go from zero to maximum level.

Decay *0...127*

Determines the decay rate or amount of time it takes for a signal to reach the **Sustain** level

Sustain *0...127*

Determines the sustain level which is held until a note ends.

Release *0...127*

Once the note has ended, the release phase begins. During this phase, the envelope fades to zero at the rate determined by the Release value.

Amplifier Envelope

This envelope is designed to control the sound volume, but can also be used for other modulations. The following parameters determine the envelope's response:

Amplifier Env

AE Attack	Decay	Sustain	Release
000	035	090	020

Attack *0...127*

Determines the attack rate or amount of time it takes for a signal to go from zero to maximum level.

Decay *0...127*

Determines the decay rate or amount of time it takes for a signal to reach the **Sustain** level

Sustain *0...127*

Determines the sustain level which is held until a note ends.

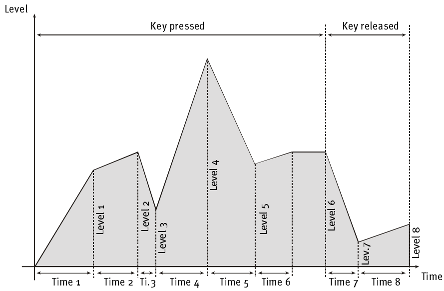
Release *0...127*

Once the note has ended, the release phase begins. During this phase, the envelope fades to zero at the rate determined by the Release value.

Wave Envelope

The microWAVE II/XT/PC's wave envelope features a multi segment characteristic with 8 separately adjustable times and levels.

ⓘ Multi segment envelopes are extremely flexible modulation sources. Their structure is made of grouped time/level parameters that allows one to generate an almost free modulation amount over several time segments. The following diagram illustrates the structure of a multi segment envelope:



As shown in the diagram, the envelope consists of several single segments. Also the figure can be divided into a sustain and a release phase. The crossover point between these two phases can be determined by selecting the corresponding segment number. The envelope is started by pressing a key. It ascends to the **Level 1** value at the rate determined by the **Time 1** parameter. In the next time segment **Time 2** the amplitude moves to the **Level 2** value. The same procedure is processed for the following segments until the end of the sustain phase is reached. In the shown example **Level 6** is the last value of the sustain phase. The amplitude remains at this value until the key is released. The envelope then moves on to process the remaining segments until it finally ends with its last value **Level 8**. In fact you can reduce the number of processed segments to get things easier. Additionally you can repeat specific segments by installing loops in the sustain phase as well as in the release phase.

Wave Env /1...4

Time 1	Level 1	Time 2	Level 2
Ø2Ø	1ØØ	115	Ø63

Wave Env /5

Key On Loop	Loop Start	Loop End

Wave Env /6

Key Off Loop	Loop Start	Loop End

Time 1...8 *0...127*
Determines the time for the individual segment to reach its end level.

Level 1...8 *0...127*
End level that the corresponding segment finally reaches.

Key On Loop *off / on*
Selects whether a loop is performed in the envelope's sustain phase or not.

Loop Start *1...8*
Defines the starting point for the sustain loop if **Key On Loop** is enabled

Loop End *1...8*
Defines the ending point for the sustain loop if **Key On Loop** is enabled. It further determines the end of the sustain phase and the beginning of the release phase. Note that this feature is also valid when **Key On Loop** is disabled

Key Off Loop *off / on*
Selects whether a loop is performed in the envelope's release phase or not.

Loop Start *1...8*
Defines the starting point for the release loop if **Key Off Loop** is enabled

Loop End *1...8*
Defines the ending point for the release loop if **Key Off Loop** is enabled. It further determines the last segment of the whole envelope. No segment beyond the selected number will be used. Note that this feature is also valid when **Key Off Loop** is disabled.

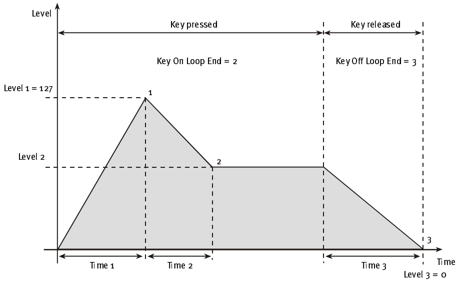
The loop points are numbered from 1 to 8. Each number represents the end of the corresponding segment, e.g. **no. 3** means the point of **Level 3** after **Time 3**. As you can see, the first loop point is at the end of segment 1. Therefore segment 1 can not be looped.

The following examples illustrate the use of the Wave Envelope:

☞ This is how you setup an classic ADSR-like envelope:

1. Set **Key On Loop** and **Key Off Loop** to ***off***. This ensures that no loops are performed.
2. Set **Level 1** to ***127***.
3. Specify the Attack time via the **Time 1** parameter.
4. Specity the Decay time via **Time 2**.
5. Use **Level 2** to setup the Sustain level.
6. Set **Key On Loop Start** to ***1*** and **Key On Loop End** to ***2***. This specifies segment 2 of the envelope as last segment in the sustain phase.
7. Set **Level 3** to ***0***.
8. Specify the Release time via **Time 3**.
9. Set **Key Off Loop End** to ***3***. This causes the envelope to stop after segment 3.

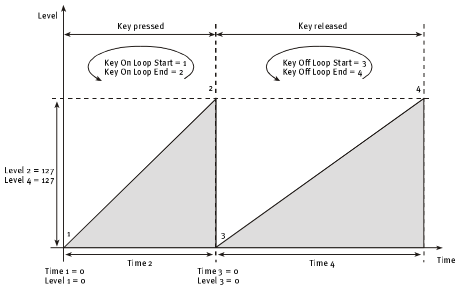
The following diagram shows how this example works:



☞ This is how you setup an envelope that it works like a sawtooth LFO with different rates in the sustain and release phase:

1. Set **Key On Loop** and **Key Off Loop** to ***on***. This causes both loops in the sustain and in the release phase to be activated.
2. Set **Level 1** and **Time 1** to ***0***. This deactivates segment 1 because it can not be looped.
3. Set **Level 2** to ***127***. This defines the maximum value of the sawtooth's amplitude.
4. Specify the rate of the sawtooth for the sustain phase via the **Time 2** parameter.
5. Set **Key On Loop Start** to ***1*** and **Key On Loop End** to ***2***. This will repeat segment 2 of the envelope as long as the key is pressed.
6. Set **Level 3** to ***0***. This defines the minimum setting of the sawtooth's amplitude.
7. Set **Time 3** to ***0***. This causes the envelope abruptly to minimum level after releasing the key and sets the minimum value of the sawtooth's amplitude in the release phase.
8. Set **Level 4** to ***127***. This defines the maximum value of the sawtooth's amplitude in the release phase.
9. Specify the rate of the sawtooth for the release phase via the **Time 4** parameter.
10. Set **Key Off Loop Start** to ***3*** and **Key Off Loop End** to ***4***. This will repeat segment 4 of the envelope in the release phase.

The following diagram shows how this example works:



Free Envelop

In addition to the previously described envelopes, the microWAVE II/XT/PC offers a Free Envelope which can be used for modulation purposes. This envelope also features a multi segment structure. It consists of 4 segments and has no loop functionality. The first 3 segments always belong to the sustain phase, the last one always belongs to the release phase. The main difference to the other envelopes is that the Free Envelope features bipolar levels. Therefore it can generate modulation amounts in the range ***-1...0...+1***.

Free Env /1

Time 1	Level 1	Time 2	Level 2
020	100	115	063

Free Env /2

Time 3	Level 3	Release	R. Level
095	070	064	025

Time 1...3 ***0...127***
Determines the time for the individual segment to reach its end level.

Level 1...3 ***-64...+63***
End level that the corresponding segment finally reaches.

Release ***0...127***
Determines the length of the release phase when the key is released. The envelope then descends to the R. Level.

R. Level ***-64...+63***
Last level that is reached when the release phase ends.

Low-frequency Oscillators (LFOs)

In addition to the main oscillators, the microWAVE II/XT/PC is equipped with two low-frequency oscillators which can be used for modulation purposes. Each LFO generates a periodic waveform with adjustable frequency and shape.

LFO 1

LFO 1 / 1

Rate	Shape	Delay	Sync
028	triangle	005	off

LFO 1 / 2

Symmetry	Humanize
027	003

Rate ***0...127 (128 Bars...1/64)***

Determines the frequency of the generated signal. If Sync is set to Clock, the value is shown in musical notation. The basis is a 4/4 beat. Triplets (e.g. ***1/8T***) and dotted notes (e.g. ***1/16.***) are available for some values.

Shape ***sine / triangle / square / sawtooth / random / S & H***

Determines the type of waveshape to be generated.

Sample & Hold samples a random value and holds it until the next LFO cycle begins. If **Rate** has a value of ***0***, then a random value is generated for each new incoming MIDI note. More variations can be achieved by means of the **Symmetry** parameter. Please read the corresponding paragraph later on in this chapter.

Delay ***off / retrigger / 1...126***

Determines the start of the LFO cycle after an incoming MIDI note.

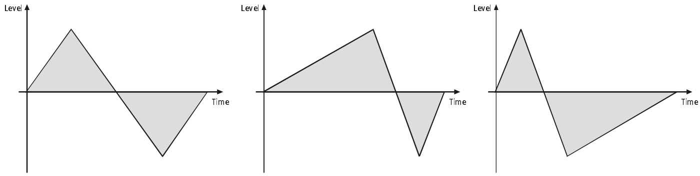
- If ***off*** is selected, the LFO runs completely free, which means its cycle is not synchronised to the note start. Use this setting e.g. when modulating the filter cutoff of a sound that should be different each time you play it.
- If ***retrigger*** is selected, the LFO starts its cycle after receiving a note. This is also known as "key sync" feature. This setting is useful when the LFO must always start at a fixed value, e.g. when creating an alert sound.
- If ***1...126*** is selected, the LFO is works like in ***retrigger*** mode, but is delayed with the specified amount. This setting is useful e.g. for solo sounds with a vibrato or tremolo that is only applied on long notes.

Sync ***off / on / Clock***

Selects if the LFO is synchronised. If off is selected, the LFO runs completely independent. If on is selected, all LFOs of the MicroWave's voices used by the sound program behave as one. If Clock is selected, the LFO is synchronised to an incoming MIDI Clock signal.

Symmetry *-64...+63*

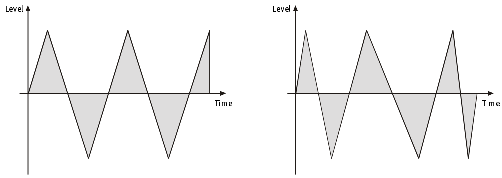
Adjusts the relationship between the rising and the falling edge of the signal. When set to 0 the generated waveshape is symmetrical. When set to positive values, the positive cycle becomes longer and the negative cycle becomes shorter and vice versa. Use this parameter to change to pulsewidth of the square signal. When using it on a triangle waveshape, you can get a sawtooth wave with a soft rising or falling slope. The following diagram illustrates this effect:



Symmetry = 0 Symmetry = +40 Symmetry = -40

Humanize *off / 1...127*

Allows one to add a random variation to the LFO speed. When disabled, the LFO remains at its initial speed, preset by the **Rate** parameter. Low settings add a human touch to the sound, high settings are useful when creating effect sounds with an irregular character e.g a wind sound where the filter frequency is modulated by an LFO. The following diagram shows the effect of the Humanize setting:



Humanize = off Humanize = 127

LFO 2

The second LFO offers the same functionality as the first one. In addition it can be linked with LFO 1.

LFO 2 / 1

Rate	Shape	Delay	Sync
028	triangle	005	off

LFO 2 / 2

Symmetry	Humanize	Phase
027	003	090

Rate *0...127*

Determines the frequency of the generated signal.

Shape *sine / triangle / square / sawtooth / random / S & H*

Determines the type of waveshape to be generated.

Sample & Hold samples a random value and holds it until the next LFO cycle begins. If **Rate** has a value of ***0***, then a random value is generated for each new incoming MIDI note. More variations can be achieved by means of the **Symmetry** parameter. Please read the corresponding paragraph later on in this chapter.

Delay ***off / retrigger / 1...126***

Determines the start of the LFO cycle after an incoming MIDI note.

- If ***off*** is selected, the LFO runs completely free, which means its cycle is not synchronised to the note start. Use this setting e.g. when modulating the filter cutoff of a sound that should be different each time you play it.
- If ***retrigger*** is selected, the LFO starts its cycle after receiving a note. This is also known as "key sync" feature. This setting is useful when the LFO must always start at a fixed value, e.g. when creating an alert sound.
- If ***1...126*** is selected, the LFO is works like in ***retrigger*** mode, but is delayed with the specified amount. This setting is useful e.g. for solo sounds with a vibrato or tremolo that is only applied on long notes.

Sync ***off / on***

Selects if the LFO is synchronised. If off is selected, the LFO runs completely independent. If on is selected, all LFOs of the MicroWave's voices used by the sound program behave as one.

Symmetry ***-64...+63***

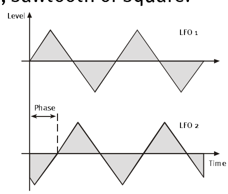
Adjusts the relationship between the rising and the falling edge of the signal. When set to 0 the generated waveshape is symmetrical. When set to positive values, the positive cycle becomes longer and the negative cycle becomes shorter and vice versa. Use this parameter to change to pulsewidth of the square signal. When using it on a triangle waveshape, you can get a sawtooth wave with a soft rising or falling slope. Please refer to the description of LFO 1 to get further information.

Humanize ***off / 1...127***

Allows one to add a random variation to the LFO speed. When disabled, the LFO remains at its initial speed, preset by the **Rate** parameter. Please refer to the description of LFO 1 to get further information.

Phase ***off / 2...180***

If disabled, LFO 2 operates independently from LFO 1. If enabled, the frequency of the generated signal is determined by LFO 1. The Phase parameter defines the angle in degrees from which LFO 2's signal is phase shifted to LFO 1. The use of this function only makes sense when using a regular waveshape like sine, triangle, sawtooth or square.



Modifiers and Modulation Matrix

The modifiers allow you to perform mathematical functions on modulation signals. Depending on the function type selected, calculation is done between two source signals or between a source signal and a constant parameter. You can use up to four independent modifier units. The result of each operation is not processed directly but can be used as input source for the modulation matrix described in the next chapter. Also you can use it again as source for another modifying process. In addition a separate delay line can be used to process a modulation source.

The following table shows an overview of all modulation sources available on the microWAVE II/XT/PC:

Setting	Description
off	Modulation off
LFO1	LFO 1 signal
LFO1*Modw	LFO 1 signal multiplied with Modwheel
LFO1*Prs.	LFO 1 signal multiplied with Aftertouch
LFO2	LFO 2 signal
FilterEnv	Filter Envelope signal
Ampl. Env	Amplifier Envelope signal
Wave Env	Wave Envelope signal
Free Env	Free Envelope signal
KeyFollow	Same as Keytrack, but with pitchbend and glide
Keytrack	MIDI note number
Velocity	MIDI note velocity
Rel. Velo	MIDI note release velocity
Pressure	MIDI aftertouch
Poly Prs.	MIDI polyphonic pressure
PitchBend	MIDI pitchbend signal
Modwheel	MIDI modulation wheel (controller #1)
Sust. Ctr.	MIDI sustain pedal (controller #64)
Foot Ctr.	MIDI foot control (controller #4)
BreathCtr.	MIDI breath control (controller #2)
Control W	Assignable MIDI-Controller 1
Control X	Assignable MIDI-Controller 2
Control Y	Assignable MIDI-Controller 3
Control Z	Assignable MIDI-Controller 4
Ctr Delay	Modifier Delay
Modify #1	Modifier #1 result
Modify #2	Modifier #2 result
Modify #3	Modifier #3 result
Modify #4	Modifier #4 result
MIDIClock	MIDI clock signal
Minimum	constant for minimum modulation (equals 0)
Maximum	constant for minimum modulation (equals +1)

Table 2: Modulation sources

Modifier Delay

This function allows one to delay a freely-definable modulation source for an adjustable period of time.

Modifier Delay

Control Delay Time	Source
Ø47	FilterEnv

Control Delay Time ***0...127***
Determines the time for which the modulation signal is delayed.

Source ***see Table 2***
Selects the modulation source whose signal is used as input for the delay line.

Modifier Units

Modifier 1...4

Source #1	Source #2	Type	Parameter
LFO1	Control X	+	Ø25

Source #1 ***see Table 2***
Selects the first source signal used for the calculation. Table 2 shows all possible settings.

Source #2 ***see Table 2***
Selects the second source signal when two sources are required for the calculation. See description of modifier functions for further details. The possible settings are the same as for Source #1.

Type ***see Table 3***

Determines which kind of operation will be performed on the selected input sources. The following types are available:

Setting	Description
+	Addition
-	Subtraction
*****	Multiplication
/	Division
XOR	Exclusive OR function
OR	OR function
AND	AND function
S & H	Sample & Hold
Ramp	Triggered ramp
Switch	Switch
abs value	Absolute value
min value	Minimum value
max value	Maximum value
lag proc.	Ramp function
filter	Low pass filter
diff.	Differential function

Table 3: Modifier functions

The result of a modifier operation always lies within the range ***-1...0...+1***. When it is assigned to a parameter in the Modulation Matrix, it is scaled to the range of the selected parameter.

The following paragraph describes the function and the result of each modifier function in detail:

+ Returns the sum of **Source #1** and **Source #2**.

- Returns the difference of **Source #1** and **Source #2**.

******* Returns the product of **Source #1** and **Source #2**.

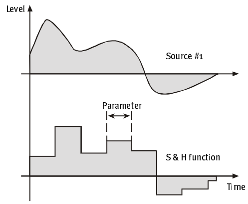
/ Returns the quotient of **Source #1** and **Source #2**.

XOR Returns the binary exclusive-or operation of **Source #1** and **Source #2**.

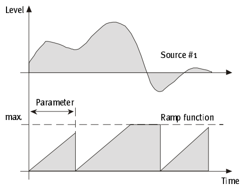
OR Returns the binary or operation of **Source #1** and **Source #2**.

AND Returns the binary and operation of **Source #1** and **Source #2**.

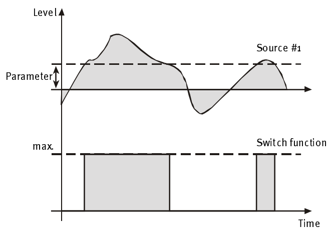
S & H Samples and holds the value of **Source #1** in regular intervals, determined by the value of **Parameter**. You can use this function to create rhythmcally modulations based on a definable source.



Ramp Creates a linear ramp from minimum to maximum. The ramp is triggered each time **Source #1** has a positive transition. The rise time is specified by **Parameter**. You can use this e.g. to get an additional sawtooth source from an LFO while another waveform is selected.



Switch Returns maximum, if the value of **Source #1** is above the value of **Parameter**. Otherwise minimum is returned. Use this function to trigger an action depending on a source signal's value. E.g. applying ring modulation when notes are played with maximum velocity. You can use this also to create a pulse signal out of an LFO, where **Parameter** determines the pulse width.

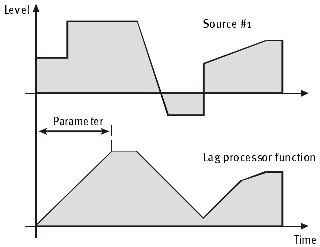


abs value Returns the value of **Source #1** without its sign. Negative values are converted to their corresponding positive amounts. **Parameter** has no function here. This function can be used e.g. for converting a bipolar modulation source to a unipolar one, like opening the filter via Pitchbend independent of the bending direction.

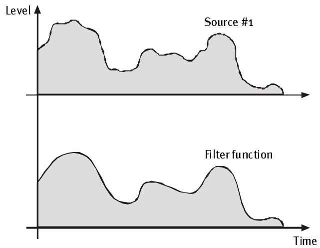
min value Returns the minimum value of either **Source #1** or **Source #2**.

max value Returns the maximum value of either **Source #1** or **Source #2**.

lag proc. The lag processor creates a linear ramp from its current value, which is initially minimum, to the value of **Source #1**. Then the ramp is stopped until **Source #1** changes again. The ramp time is specified by **Parameter**. This function is useful when you want to apply a definable modulation over a specified time, e.g. Modwheel controlled ramp for oszillator sweeps.



filter Performs a low pass filter function on **Source #1**. The filter frequency is determined by **Parameter's** value. Use this function to smooth a signal.



diff. Performs a differential function on **Source #1**. The result of this function represents the speed of value change in the selected source. **Parameter** has no function here. This function is useful to detect if a source signal has changed, e.g. the Modwheel was turned.

Parameter ***0...127***

Defines a value for modifier functions that require a constant parameter. See the **Type** parameter described above for further details.

Modulations matrix

A modulation can be described as influencing a sound parameter by a signal generating unit. The terms used in this context are "source" and "destination". The microWAVE II/XT/PC offers 16 independent modulation assignments each with individual settings of source, destination and amount.

Mod 1...16

Source	Amount	Destination [5]
Modwheel	+047	Wave1 Pos

Source ***see table 2***

Defines the modulation source. See Table 2 for the list of available sources.

Amount ***-64...+63***

Determines the amount of modulation applied to the destination. Since the modulation is in fact a multiplication of the source signal and this parameter, the resulting amplitude depends on the type of modulation source you select:

- For the so-called unipolar modulation sources, the resulting amplitude lies within the range of ***0...+1***, if Amount is positive or ***0...-1***, if Amount is negative. These sources are: Filter Envelope, Amplifier Envelope, Wave Envelope, all MIDI controllers including Modwheel, Foot control etc., Velocity, Release Velocity, Aftertouch, Polyphonic Pressure and MIDI clock.
- •For the so-called bipolar modulation sources, the resulting amplitude lies within the range of ***-1...0...+1***. These sources are: Free Envelope, both LFOs, Keytrack, Keyfollow and Pitchbend.

For the modulation sources Keytrack and Keyfollow, a value of ***+56*** represents 100% of the scale.

Destination ***siehe Tabelle 4***

Defines the modulation destination. The table below shows all possible settings for this parameter:

Setting	Description
Pitch	Global pitch off all oscillators
Osc1 Pit.	Oscillator 1 pitch
FM Amount	Amount of frequency modulation
Osc2 Pit.	Oscillator 2 pitch
Wave1 Pos	Wave 1 startposition
Wave2 Pos	Wave 2 startposition
Wave1 Mix	Mixer input level Wave 1
Wave2 Mix	Mixer input level Wave 2
Ringmod	Mixer ringmodulation level
Noise Mix	Mixer noise level

Table 4: Modulation destinations

Setting	Description
Cutoff	Filter 1 cutoff frequency
Resonance	Filter 1 resonance
Filter 2	Filter 2 cutoff frequency
Volume	Amplifier master volume
Panning	Amplifier pan position
FE Attack	Filter Envelope attack
FE Decay	Filter Envelope decay
FE Sustain	Filter Envelope sustain
FE Release	Filter Envelope release
AE Attack	Amplifier Envelope attack
AE Decay	Amplifier Envelope decay
AE Sustain	Amplifier Envelope sustain
AE Release	Amplifier Envelope release
WE Times	All Wave Envelope times
WE Levels	All Wave Envelope levels
Free Env T	All Free Envelope times
Free Env L	All Free Envelope levels
LFO1 Rate	LFO 1 rate
LFO1 Level	LFO 1 level
LFO2 Rate	LFO 2 rate
LFO2 Level	LFO 2 level
M1 Amount	Amount of modulation assignment 1
M2 Amount	Amount of modulation assignment 2
M3 Amount	Amount of modulation assignment 3
M4 Amount	Amount of modulation assignment 4

Table 4: Modulation destinations

Program Name

This page is designed to name the Sound program. You can use up to 16 characters for this purpose.

Name

Position	Character		
Ø1	U	Unisono	WMF

First select the character to be modified via the first value dial. Then change its setting via the second value dial.

In the microEd!t software, naming a sound program is much easier. Just type in a name with the PC keyboard and press the [Return] key.

MULTI MODE

MULTI PARAMETERS

The Multi parameters consist of settings which are common to all instruments in a multi program.

Note: you'll find further information about programming the multi-mode in the Quick Start on page 12.

Volume

Multi Volume 127

Tempo

Multi Arpeggiator Tempo 130

Controls

Control W 004	Control X 008	Control Y 011	Control Z 012

Name

Position 01	Character M MIDI Multi

Multi Volume *0...127*
Determines the master volume for the multi program.

Arpeggiator Tempo *extern / 50...300*
This setting allows one to define a master tempo for all instruments in the multi program. If extern is selected, the tempo is determined by MIDI clock.

Control W...Control Z *0...120 / global*
These parameters are used to define modulation sources that are freely definable MIDI controllers. Each value represents a MIDI controller number that is used when you assign its parameter as modulation source in the Modifiers or the Modulation Matrix. If ***global*** is selected, the corresponding settings made in the global parameter section are used.

Name
Use this page to set the multi program's name. First select the character to be modified via the first value dial. Then change its setting via the second value dial.

Instrument parameters

The Instrument parameters consist of individual settings for each Instrument in a multi program.

Selecting an instrument for editing

Before you apply any edits to an Instrument's parameter, you have to select the Instrument to which the edits belong. Use the rightmost value dial l to switch between the Instruments.

Instrument Select (e.g. 1)

Bank	Sound Unisono WMF
A	AØØ1 Inst. #1

The instrument no. is always displayed when a parameter page with Instrument relating settings is selected. This is also valid when editing a sound program in Multi Mode because the sound program belongs to an Instrument. The no. is not displayed while editing Multi or Global parameters.

When editing an Instrument's Sound program, you can also switch among the Instruments by turning the rightmost value dial l when the **Shift** key ⓚ is hold.

Sounds

Sound 1

Bank	Sound Unisono WMF
A	AØØ1 Inst. #1

Sound 2

Channel	Volume	Status
Ø5	Ø9Ø	on Inst. #1

Sound 3

Panning	PanMod	Output
center	normal	Main Out Inst. #1

Bank ***A / B***
Selects the bank from which the sound program is taken.

Sound ***001...128***
Selects the instrument's sound program.

Channel *global / omni / 1...16*

Determines the MIDI receive channel for the instrument.

- If ***omni*** is selected, the Instrument receives on all channels.
- If ***global*** is selected, the MIDI channel defined in the global parameters is used.

Volume *0...127*

Determines the master volume for the instrument.

Status *off / on*

Determines whether the instrument is disabled or enabled.

Panning *left 64...center...right 63*

Determines the position of the instrument within the stereo panorama. The value range extends from ***left 64***, which means far left, over the ***center*** position to ***right 63***, which means far right.

PanMod *off / normal / inverse*

This setting decides whether panning modulation is applied or not.

- When set to ***off***, no panning modulation is done at all.
- When set to ***normal***, panning modulation is applied as defined in the single program that is used for the instrument.
- When set to ***inverse***, panning modulation is done as before, but the modulation signal is negated and, as a result, the stereo sides are exchanged.

Output *Main Out / Sub Out*

Selects the audio output on which the instrument's signal will appear. Main routes the instrument to the main outputs **Main Out Left/Stereo** and **Main Out Right Mono**, ***Sub*** routes it to the sub outputs **Sub Out Left/Stereo** and **Sub Out Right Mono**.

TUNE

Tune

Transpose	Detune	
Ø12	+ØØ	Inst. #1

Transpose *-48...+48*

Allows one to transpose the instrument in steps of a semitone.

Detune *-64...+63*

Fine-tunes the instrument in increments of 64ths of a semitone.

Range

Range 1

Lowest	Highest Velocity	
001	063	Inst. #1

Range 2

Lowest	Highest Key	
000	127	Inst. #1

Lowest Velocity *1...127*
This parameter allows you to limit the velocity range in which the instrument is played. Only notes with a velocity higher or equal to the selected value are passed through. Set this parameter to *1*, if you want to turn velocity switching off.

Highest Velocity *1...127*
Counterpart to the **Lowest Velocity** parameter. Only notes with a velocity lower or equal to the selected value are passed through. Set this parameter to *127*, if you want to turn velocity switching off.

Lowest Key *0...127*
Equivalent to the velocity switching parameters, you can restrict the key range used for the instrument's tone generation. Only notes with a key number higher or equal to the selected value are passed through. Set this parameter to *0* if you want to use the full keyboard range.

Highest Key *0...127*
Counterpart to the **Lowest Key** parameter. Only notes with a key number lower or equal to the selected value are passed through. Set this parameter to *127* if you want to use the full keyboard range.

Arpeggiator

Every Instrument in a Multi mode program is capable of using its own arpeggiator. The settings made in this section override the settings defined in the Instrument's Sound program. All Instruments will use the tempo setting defined in the **Multi Arpeggiator Tempo** parameter, because it makes no sense to use different settings for each Instrument. Alternatively, you can use the original settings of the Sound program by using the corresponding option in the Active parameter.

Arpeggiator 1

Active	Clock	Range	
Sound Arp	1/2	02	Inst. #1

Arpeggiator 2

Pattern	Direction	Note Order	
off	up	by note	Inst. #1

Arpeggiator 3

Velocity roote note	Reset on Pattern Start off Inst. #1

Active ***off / on / hold / Sound Arp***

Enables or disables the arpeggiator or activates the hold mode. When ***hold*** is activated, incoming MIDI chords generate continuous arpeggios even when the chord is released. If ***Sound Arp*** is selected, the arpeggiator uses the settings defined in the Sound program that builds the instrument.

Clock ***1/1...1/32***

Determines the note value for whole notes to thirty-second notes. The basis is a 4/4 beat. Triplets (e.g. ***1/8T***) and dotted notes (e.g. ***1/16.***) are available for every value.

Range ***1...10***

Determines the range of the single notes in octaves.

Pattern ***off / user / 1...15***

Determines whether an rhythm pattern is played and which one.

- If ***off*** is selected, the arpeggiator playes its notes in regular steps, specified by the **Clock** parameter.
- If ***user*** is selected, the arpeggiator uses the free programmable pattern defined in the ***Arpeggiator User Pattern*** page of the sound program. The instrument itself does not provide a user pattern.
- Additionally, the arpeggiator features 15 preset rhythm patterns. These are numbered from ***1*** through ***15***.

See diagram 4 in chapter "Sound Parameters" to get detailed information about patterns.

Direction ***up / down / alternate / random***

Determines the sequence of generated notes according to pitch.

- If ***up*** is selected, the arpeggio starts at the lowest note and sweeps up through the notes until it reaches the highest note. It then starts at the bottom again.
- If ***down*** is selected, the arpeggio starts at the highest note and sweeps down through the notes until it reaches the lowest note. It then starts at the top again.
- If ***alternate*** is selected, the arpeggio starts at the lowest note and sweeps up through the notes until it reaches the highest note. It then starts to sweep back down.
- If ***random*** is selected, the arpeggio plays any of the notes in a random order.

NoteOrder ***by note / note rev. / as played / reversed***

Determines the sequence of generated notes according to note order.

- If ***by note*** is selected, the arpeggio sequence is sorted by the MIDI note number. This is the standard mode, used by most arpeggiators.
- If ***note rev.*** is selected, the arpeggio sequence is sorted in the exactly reversed order to the ***by mode*** setting.
- If a***s played*** is selected, the arpeggio is generated in the order of the incoming notes. In combination with the user programmable pattern this feature offers a small but effective step sequencer.
- If ***reversed*** is selected, the arpeggio is generated in the reverse order of the incoming notes.

To understand the difference of the individual settings, it is nessessary to "step-input" the notes of the chord as described in the chapter "Arpeggiator" of the sound parameters.

Velocity ***root note / last note***

Determines how the velocity values of the generated notes are calculated.

- If ***root note*** is selected, every generated note inherits its velocity from its base note. E.g. if the base chord for the arpeggio contains an E with a certain velocity, all generated E notes also have this velocity value, independent of their octave setting.
- If ***last note*** is selected, every generated note has the same velocity as the last incoming note.

Reset on Pattern Start ***off / on***

Selects if the arpeggiator is reset each time the rhythm pattern starts again. If the setting is disabled, the arpeggiator plays all chord notes from the first to the last and over again, regarding the sequence determined by ***Direction*** and ***Note Order***. If the setting is enabled, the arpeggiator only plays the number of chord notes that correspond to the pattern length. Then it starts with the first chord note at its basic octave again. The result is similar to pressing the chord again each time the pattern restarts.

Global Parameters

Global parameters are settings that influence the microWAVE II/XT/PC's general response. These are determined separately from the programs and stored in a special memory location. Global parameters are stored automatically when you modify them, so you are not required to save them separately.

MIDI 1

Channel 12	PrgChange multi	BendRange 012	Device ID 000

MIDI 2

Parameter Control	Send Ctl+SysEx	Receive on

MIDI 3

MIDI Clock	Send off	

Controls

Control W 004	Control X 008	Control Y 011	Control Z 012

Volume

Main Volume 100	Input Gain 2

Tune

Master Tuning 440 Hz	Transpose +00

System

Display timeout 064	Contrast 100

Channel ***omni / 1...16***

Sets the basic send and receive channel for the microWAVE II/XT/PC. This setting is valid for all Sound programs and for Instruments of a Multi program whose **Channel** parameter is set to ***global***. If ***omni*** is selected, the microWAVE II/XT/PC sends on channel 1 and receives on all channels.

PrgChange ***sound / multi / combined***

Determines the way MIDI Program Change messages are processed.

- If ***sound*** is selected, program changes are used to select Sound programs for the Instrument that receives on the corresponding MIDI channel.

- If ***multi*** is selected, the whole Multi program is switched by program changes, that are received on the basic **channel** set above.
- If ***combined*** is selected, Instrument programs can be changed by using the Instrument's channel, the Multi can be changed by using the basic **channel**.

BendRange ***0...120 / harmonic***

Determines the intensity of the pitchbend via MIDI Pitchbend messages in semitones. If ***harmonic*** is selected, the pitchbend is performed in steps of the harmonic and subharmonic scale. Please refer to the chapter "Oscillator" to get further information about the harmonic scale. This setting is valid for all programs whose oscillator **Pitchbend Range** parameter is set to ***global***.

Device ID ***0...126***

Defines the device identification number for system exclusive data transmission. Transmission will only be executed successfully if the sender and receiver setting coincide. Device ID ***127*** is a so-called broadcast ID that addresses all connected microWAVE II/XT/PCs. The microWAVE II/XT/PC can receive this from other devices, but cannot send it itself. This function is limited to special computer software.

Par. Control Send ***off / Ctl only / SysEx / Ctl+SysEx***

ⓘ This parameter has no effect in microWAVE PC.

Par. Control Receive ***off / on***

Enables or disables the receiving of parameter control messages via MIDI. These messages include controller and system exclusive data.

MIDI Clock Send ***off / on***

ⓘ This parameter has no effect in microWAVE PC.

Control W...Control Z ***0...120***

These parameters are used to define modulation sources that are freely definable MIDI controllers. Each value represents a MIDI controller number that is used when you assign its parameter as modulation source in the Modifiers or the Modulation Matrix. The settings made here are only valid for Sound programs because each Multi program has its own set of Control W...Control Z parameters.

☞ Example: You want to control the LFO1 speed via MIDI controller #49. To do so, set **Control W** to ***49*** first. Then, setup an entry in the Modulation Matrix of your sound program with **Control W** as source and ***LFO1 Rate*** as destination and apply an suitable amount. In the same way you can use Control X...Control Z for further assignments.

Main Volume ***0...127***

Adjusts the master volume of all microWAVE II/XT/PC's programs on both outputs. This setting is also accessible from the Play page.

Input Gain ***1...4***

Sensitivity of the external audio input.

Master Tuning ***430...450 Hz***

Determines the microWAVE II/XT/PC's overall pitch. The value specified here is the reference pitch for MIDI note A3. The default setting is 440Hz, which is commonly used by most instruments.

You should only change this setting if you really know what you're doing. You will have to adjust all your other instruments, too. Don't forget to set it back again!

Transpose ***-12...+12***

Allows one to set a global pitch transpose for all programs of the microWAVE II/XT/PC.

Display timeout ***0...127***

Determines how long the page names are displayed in the upper right corner when calling a parameter page via the page dial C. You may want to decrease the value or set it to ***0*** after you have got some experience with the microWAVE II/XT/PC.

Contrast ***0...127***

Sets the display contrast (only microWAVE II/XT).

MIDI CONTROL

This chapter describes the options you have available to control the microWAVE II/XT/PC via MIDI.

SELECTING PROGRAMS

All of the microWAVE II/XT/PC's Sound and Multi programs can be called via MIDI Program Change messages and MIDI Bank Select messages. As the device contains 128 programs in each bank, it recognizes program number ***0...127***. To select the bank, you have to use a Bank Select message:

- Bank 0 contains Sound Programs ***A001...A128***
- Bank 1 contains Sound Programs ***B001...B128***

When the microWAVE II/XT/PC is in Multi mode, you have three options, how Program Change and Bank Select messages work. By means of the Global parameter **PrgChange** you can determine if a Sound program inside the current Multi Program is changed, the whole Multi program is changed, or if both methods are used in combination.

INFLUENCING SOUNDS VIA MIDI MESSAGES

CONTROLLERS AS MODULATION SOURCES

The controllers Modwheel and Breath Control are always used as modulation sources. The freely-definable **Control X...Z** can also be used as a modulation source. X...Z stands for definable controller numbers ***1...120***. Use these controllers in the Modifiers and the Modulation Matrix.

CHANGING SOUND PARAMETERS VIA CONTROLLERS

Every important parameter is assigned a controller number through which the parameter can be changed. If a parameter is changed at the device, then this change is sent along with the appropriate controller number via MIDI. This is especially helpful when you want to record changes you made at the microWAVE II/XT/PC to a sequencer.

All controller messages are sent and received via the channel defined in the global parameters or, if in Multi mode, selected for the corresponding Instrument. The appendix of this manual contains a table listing the controller numbers and the sound parameters they are assigned to.

PITCHBENDING

The **Pitchbend Range** parameter of the oscillators lets you define to what extent a pitchbend message influences the pitch of the microWAVE II/XT/PC. Pitchbend is also available as a modulation source.

Aftertouch and Poly Pressure

Aftertouch and Poly Pressure are available as modulation sources in the microWAVE II/XT/PC. They can be used for any application where control change messages are accepted.

System Exclusive Data

All parameters of the microWAVE II/XT/PC can be controlled by system exclusive data. You can find a detailed description of the commands and data formats in the appendix.

System Exclusive Data Transmission

System exclusive data transmission lets you send and receive the contents of the microWAVE II/XT/PC's memory via MIDI (dump).

Sending System Exclusive Data

When you activate the send functions, the microWAVE II/XT/PC sends the contents of its memory to the selected MIDI output device (driver) from the Preference menu. Using a sequencer, you can record and archive this data.

Receiving System Exclusive Data

You are not required to activate a special receive mode at the MicroWave in order to receive system exclusive data via MIDI. The transmission is activated via a Dump Request command originating at the device that is sending the messages. However there are a few things you should check prior to the transmission:

- Check out the parameter **Device ID**. Data transmission will only be executed successfully if the sender and receiver setting coincide.
- Make sure none of the microWAVE II/XT/PC's programs are in Edit mode. All edit buffers are cleared via data transmission and therefore all edits that were not stored prior to the dump will be irretrievably lost!

After activating the dump command at the sender device, the microWAVE II/XT/PC will receive data and store these in its memory.

ⓘ When the microWAVE II/XT/PC receives a Sysex dump with the device ID ***127***, it will always accept the dump, regardless of the setting of its **Device ID** parameter. Device ID ***127*** is a so-called "Broadcast ID" that addresses all connected microWAVE II/XT/PCs. The microWAVE II/XT/PC can receive this from other devices, but it cannot send a Broadcast ID to other devices. This function is limited to special computer software.
Also a checksum of ***127*** is always accepted as valid.

Appendix

MIDI Controller Assignments

Contr. No.	Range	Parameter	Value Range
1	0...127	Modulation wheel	0...127
2	0...127	Breath control	0...127
4	0...127	Foot controller	0...127
5	0...127	Glide Time	0...127
7	0...127	Channel Volume	0...127
10	0...127	Panning	left 64...center...right 63
12	0...1	Chorus	0:off 1:on
14	0...127	Filter Env Attack	0...127
15	0...127	Filter Env Decay	0...127
16	0...127	Filter Env Sustain	0...127
17	0...127	Filter Env Release	0...127
18	0...127	Amp Env Attack	0...127
19	0...127	Amp Env Decay	0...127
20	0...127	Amp Env Sustain	0...127
21	0...127	Amp Env Release	0...127
22	0...3	Glide Type	0:portamento 1:fingered port. 2:glissando 3:fingered gliss.
23	0...1	Glide Mode	0:exp. 1:linear
24	0...127	LFO1 Rate	0...127
25	0...5	LFO1 Shape	0:sin 1:tri 2:square 3:saw 4:random 5:S&H
26	0...127	LFO2 Rate	0...127
27	0...127	LFO2 Delay	0:off 1:retrigger 2...127:1...126
28	0...5	LFO2 Shape	0:sin 1:tri 2:square 3:saw 4:random 5:S&H
29	0...2	Filter Env Trigger	0:normal 1:single 2:retrigger
30	0...127	LFO1 Delay	0:off 1:retrigger 2...127:1...126
31	0...2	Amp Env Trigger	0:normal 1:single 2:retrigger
32	0...1	Bank Select	0:Bank A 1:Bank B
33	0...8	Osc 1 Octave	-4...+4
34	0...24	Osc 1 Semitone	-12...+12
35	0...127	Osc 1 Detune	-64...+63
36	0...121	Osc 1 Pitchbend Scale	0...120:semitones 121:harmonic
37	0...127	Osc 1 Keytrack	-100%...+200%
38	0...8	Osc 2 Octave	-4...+4
39	0...24	Osc 2 Semitone	-12...+12
40	0...127	Osc 2 Detune	-64...+63
41	0...1	Osc 2 Sync	0:off 1:on
42	0...121	Osc 2 Pitchbend Scale	0...120:semitones 121:harmonic

Table 5: MIDI Controller Assignments

Contr. No.	Range	Parameter	Value Range
43	0...127	Osc 2 Keytrack	-100%...+200%
44	0...1	Osc 2 Link	0:off 1:on
45	0...127	Wave 1 Level	0...127
46	0...127	Wave 2 Level	0...127
47	0...127	RingMod Level	0...127
48	0...127	Noise Level	0...127
50	0...127	Filter 1 Cutoff	0...127
51	0...127	Filter 1 Keytrack	-200%...+197%
52	0...127	Filter 1 Env Amount	-64...+63
53	0...127	Filter 1 Env Velocity	-64...+63
54	0...5	Filter 1 Type	0:24dB LP 1:12dB LP 2:24dB BP
55	0...127	Amp Keytrack	-200%...+197%
56	0...127	Filter 1 Resonance	0...127
57	0...127	Amp Volume	0...127
58	0...127	Amp Env Velocity	-64...+63
60	0...127	Filter 2 Cutoff	0...127
61	0...1	Filter 2 Type	0:6dB LP 1:6dB HP
62	0...127	Filter 2 Keytrack	-200%...+197%
64	0...127	Sustain Switch	0...127
65	0...127	Glide on/off	0...127
70	0...127	Wavetable	Wavetable 001...128
71	0...63	Wave 1 Startwave	00...60 61:triangle 62:square 63:saw
72	0...127	Wave 1 Phase	0:free 1...127:3°...357°
73	0...127	Wave 1 Env Amnt.	-64...+63
74	0...127	Wave 1 Env Vel. Amnt.	-64...+63
75	0...127	Wave 1 Keytrack	-200%...+197%
76	0...1	Wave 1 Limit	0:off 1:on
77	0...63	Wave 2 Startwave	00...60 61:triangle 62:square 63:saw
78	0...127	Wave 2 Phase	0:free 1...127:3°...357°
79	0...127	Wave 2 Env Amnt.	-64...+63
80	0...127	Wave 2 Env Vel. Amnt.	-64...+63
81	0...127	Wave 2 Keytrack	-200%...+197%
82	0...1	Wave 2 Limit	0:off 1:on
83	0...1	Wave 2 Link	0:off 1:on
85	0...127	Free Env Time 1	0...127
86	0...127	Free Env Level 1	-64...+63
87	0...127	Free Env Time 2	0...127
88	0...127	Free Env Level 2	-64...+63
89	0...127	Free Env Time 3	0...127
90	0...127	Free Env Level 3	-64...+63
91	0...127	Free Env Release Time	0...127
92	0...127	Free Env Release Level	-64...+63

Table 5: MIDI Controller Assignments

Contr. No.	Range	Parameter	Value Range
93	0...2	Free Env Trigger	0:normal 1:single 2:retrigger
102	0...2	Arp Active	0:off 1:on 2:hold
103	0...9	Arp Range	1...10 Octaves
104	0...15	Arp Clock	1/1...1/32
105	0...127	Arp Tempo	0:external 1...127:50...300BPM
106	0...3	Arp Direction	0:up 1:down 2:alternate 3:random
107	0...16	Arp Pattern	0:off 1:user 2...16:Pattern 1...15
108	0...3	Arp Note Order	0:by note 1:note rev 2:as played 3:reversed
109	0...1	Arp Velocity	0:root note 1:last note
110	0...1	Arp Reset	0:off 1:on
111	0...15	Arp Pattern Length	1...16
112	0...3	LFO 1 Sync	0:off 1:on 3:Clock
113	0...127	LFO 1 Symmetry	-64...+63
114	0...127	LFO 1 Humanize	0...127
115	0...3	LFO 2 Sync	0:off 1:on 3:Clock
116	0...127	LFO 2 Symmetry	-64...+63
117	0...127	LFO 2 Humanize	0...127
118	0...127	LFO 2 Phase	0:free 1...127:3°...357°
120	0	All Sound Off	
121	0	Reset All Controllers	
123	0	All notes off	

Table 5: MIDI Controller Assignments