

# the fish fillets

## channel insert VST plugins package



## USER'S MANUAL

Date: 2002-12-20

Author: Sascha Eversmeier, Berlin, Germany

URL: <http://www.digitalfishphones.com>

Email: [sascha@digitalfishphones.com](mailto:sascha@digitalfishphones.com)

Subject:

**BLOCKFISH**, dynamics compressor VST plugin, 'blockfish.dll' (PC version), 'blockfish' (MAC version)

**SPITFISH**, de-esser VST plugin, 'spitfish.dll' (PC version), 'spitfish' (MAC version)

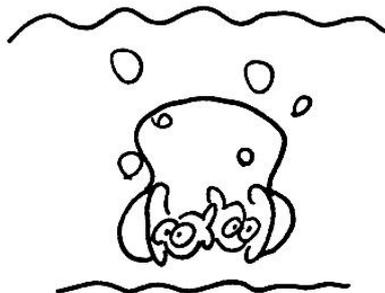
**FLOORFISH**, expander/gate VST plugin, 'floorfish.dll' (PC version), 'floorfish' (MAC version)

Current program versions: 1.0

MacOS version of this plugin compiled by Urs Heckmann (<http://www.u-he.com>)

*This manual describes the concepts behind the **fish fillets**, the functions of each plugin and the basic steps on how to use the software.*

*The **'fish fillets' package** is freeware and therefore free of charge. The latest version is always available at the author's website.*



[digitalfishphones.com](http://digitalfishphones.com)

## What's in the package?

The fish fillets package consists of three plugins for essential dynamics processing:

1. The BLOCKFISH compressor is a dynamics compressor capable of a compact and dense sound (hence the name). But unlike a lot of digital compressors, it does not 'squash' your signal that much. The focus was to keep your music alive and let it breathe. It comes with a typical 'vca' circuit providing modern compression - best for drums and guitars - and a classical 'opto' unit that recreates the sound of old photo-electrical gain devices and has been optimised for vocal tracks. The basic parameters are easily set up but you have access to further adjustments that can turn BLOCKFISH into a real 'killer' device.
2. SPITFISH serves as an easy-to-use de-esser. If you ever had problems with loud sibilants and s-like vocals, this device can help to make those nasty sounds much smoother. You can also apply it to other signals that might need some evening out within the upper frequency range, e.g. cymbals and crashes. The whole character has been tuned to 'nice' and 'gentle'. It should be almost impossible to make your recordings sound unnatural within the normal operational range.
3. FLOORFISH is a classical gate/expander unit that was originally designed to make acoustic drum recording much cleaner. Especially seperately miked snares and toms often unveil a large amount of 'bleeding noise' if you compress them. The FLOORFISH can help to lower this noise floor. Unlike a typical 'noise gate', the signal is not cut off completely if you do not want it to.

## System requirements

Each of the fish fillets is a real-time VST plugin. The package is available for the PC/Windows platform as well as for MacOS. The requirements in particular:

PC/Windows:

You will need a PC machine equipped with Windows 9x, 2000 or XP, with reasonable speed for real-time audio applications. The minimum CPU power should be 200Mhz (one instance), more is better as the CPU power consumption increases with more instances you are willing to open.

A VST-compatible software host is required, such as Steinberg Cubase VST, Emagic Logic Audio, Orion from Sonic Syndicate/Synapse or hosts that are equipped with VST-to-DirectX adapters like Samplitude 6. The plugins have been tested with the above applications. There may be others which work as well, but you will have to find out for yourself.

*(Unless not stated otherwise on the website, usage of the fish fillets - especially BLOCKFISH - with Cubase SX (with WinXP) should be considered experimental for the moment.)*

MacOS:

In order to run the Mac version of the fish fillets, any modern G3 or G4 machine should work. As with the PC version, a higher CPU speed generally means a better performance and having more instances at hand.

Any plugin of the fish fillets have been tested with Emagic Logic under MacOS 9. If Problems with other hosts on this platform occur, please let me know.



digitalfishphones.com

## Installation

That's pretty easy. As you are reading this manual, I assume that you have already extracted the archive (thereby using WinZip or a similar application on the PC or unstuffed it on the Mac). Its contents are

- Three plugin files
  - PC: 'blockfish.dll', 'spitfish.dll', 'floorfish.dll'
  - Mac: 'blockfish', 'spitfish', 'floorfish' (either MacOS 9 or MacOS X files, depending on the package)
- The user's manual (yes, which you are currently reading).

To install the plugin files, simply locate the folder named 'vstplugins' of your host program and copy the files right into it. With a lot of host programs, you can create a sub-folder that will be scanned, too. Mac OS X users should choose the directory ' ~/Library/Audio/Plug-Ins/VST/'.

That's it.

Now (re-)start the host. It will scan the appropriate folder and collect all plugins. When loading is completed, you should find each of the fish fillets within the list of available 'insert' plugins.

What are 'inserts'?

Every fish fillet is a dynamics tool, so it makes no sense to pass only a portion of the signal to it. Dynamic effects devices like this are made to process the entire signal to be as effective as possible.

Each of these plugins comes with its own user interface. You should already be familiar with your host software so that you know how to open plugins and their interfaces/editors.



## BLOCKFISH channel compressor



BLOCKFISH is a versatile dynamics compressor for single signal processing such as guitars, bass, vocals, drums or separated drum signals like kick drum or snare.

Unlike some software compressors that offer a large amount of controls and are difficult to set up appropriately, BLOCKFISH comes with only some basic (visible) controls but is capable to deliver a broad operational spectrum.

But if you want to, this plugin offers exceptional features usually not found with ordinary compressors. It is not only 'compressing' the sound; you can use it as a sound designer's tool in a creative way. For doing so, BLOCKFISH lets you make adjustments on the circuit board by turning trim pots that control directly certain sub-

function deep down. This is intended to be an 'advanced feature' given into the hands of the more experienced user. The novice is provided with basic presets to start with. Those contain appropriate settings of the circuit, accessible via switchable presets.

There's a certain philosophy behind the BLOCKFISH: don't sound like others; offer variation.

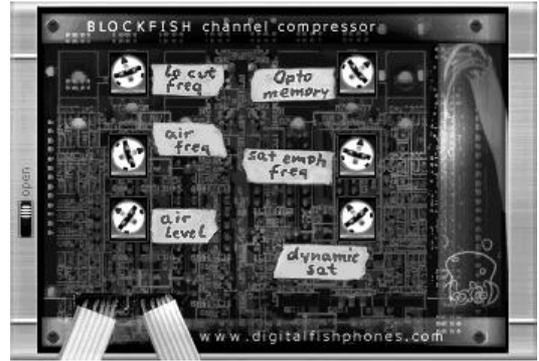
I didn't have a good compressor for vocals. All the ones were reacting much too 'digital', making the singer's voice sound squashed and lifeless. I wanted the BLOCKFISH to be able to keep the perceived level under control as well as to let the signal cut through in the mixdown. This is often a property I miss with a lot of digital compressors.

Here's a brief overview on the features of BLOCKFISH:

- 'one-knob compression': a single knob adjusts the amount of compression as well as the signal intensity for easy setting up. Compared to standard compressors, this dial means 'threshold' as well as 'ratio'. Both parameters are not exactly defined with this design. We'll come to this topic later on...
- Furthermore, the 'compression' knob automatically sets the amount of makeup gain that is to be expected with the dialled-in compression.
- Easy adjustment of the compression envelope (attack & sustain): from fast to slow response, one knob does the entire job.
- Two compression models taken from the analog realm: vca and opto-eletric circuits. The vca mode delivers that typical modern type of deep compression while following a soft transition curve from 'uncompressed' to 'fully compressed'. The opto mode has a transition that follows the response of classical opto-electric devices like a vactrol element. Compared with the vca setting, it is able to sound more 'open', but includes a small overshoot on loud signals. As with the old vintage gear, this mode is less perfect, but usually a very musical setting.
- Adaptive envelope adjustment (opto mode). In this mode, the 'response' time of the circuit is partly controlled by the signal itself. This is originally an 'artifact' of a photo-electrical element as such a device has a natural 'lag' with fast transients and stores the signal energy for some time. This has the effect of a very natural compression since the action is not static. The opto-electrical element's 'time lag' can be controlled via the 'memory' trim pot on the circuit board view.
- Low-cut filter for the detection circuit. Often with complex sources like a drum track or a bus signal, a compressor reacts mainly to the low-frequency content and produces a 'pumping' sound; the bass controls the whole spectrum. The filter cuts off that bass and helps to achieve a clean tracking and obtain maximum transparency. The circuit board view lets you alter the cutoff frequency.



- 'air' mode. Sometimes, heavy compression can make a signal sound dull. The air switch activates a circuit that adds dynamically a slight treble boost by the amount of the current gain reduction. This technique can help to refresh a signal. Doing heavy compression, the circuit adds slightly more treble than originally existed. This can sound very pleasant with vocals and is often referred as 'gloss'.
- Variable makeup-/output stage. Although the compression of BLOCKFISH includes an 'auto-makeup' feature, the output lets you compensate for any volume loss or just boost the level by up to 6dB of loudness. The unit will never clip. An analog-style saturation stage takes care of it while adding extra harmonics with loud output settings.
- Variable saturation stage. You can choose how much of analog-style saturation is introduced to the signal. The frequency response and the type of saturation can also be adjusted (on the circuit board via trim pots).
- 'Complex mode'. The 'complex' switch on the interface is a unique feature, though the concept behind it is not: this serializes two compressor stages (with saturation stage in-between). Running two compressors in series is a common engineers' trick to achieve deep compression while avoiding the typical negative artifacts usually associated with the same amount of compression on a single-compressor device.



## The BLOCKFISH topology

Consider the plugin as two separate compressing units:

One is an **opto-electrical device** which was modelled after the early compressors that consisted of a photo resistor and a light source that - when coupled - formed a regulating circuit if the light source was driven by the incoming signal in some way. Usually, the input for the light source was taken from the output of the compressor stage. This is known as 'feed-back' design and a simple and efficient approach to keeping the circuit in a stabilized state. It sounds very natural with vocals and instruments, but the feedback structure and the inertia of the opto-electrical gain control element does forbid a fast reaction.

The other unit is functioning in the same way like today's **VCA (Voltage Controlled Amplifier)** gear. Here, the gain control element consists of a circuit that follows the 'feed-forward' design: a 'detector' takes a part of the incoming signal, rectifies the waveform and calculates some sort of average level.

Common VCAs are very fast in their response behaviour and usually deliver a much deeper compression than an opto compressor, partly because of the missing feedback structure.

You might ask 'What should I use for what type of signal?'. For fast response, I'd recommend the vca setting, as well as for compression that should be very audible.

If you'd like to have a more open and smooth sound, disregarding a quick 'attack', consider using the opto setting. It's mainly been designed and optimised for vocals but might also suit basses and guitars.

The compressor circuit is followed by a saturator stage that can handle more than one task at a time:

- increasing the overall loudness
- introducing analog-style saturation products by means of harmonic distortion. This is similar to driving a tape into the 'red' region
- compensating some part of the current gain reduction by turning the 'dynamic sat' knob
- limiting transients. The more saturation, the more 'natural' limiting occurs

The 'real' topology of BLOCKFISH is much more complicated and highly interactive to the adjustments as well as the signals influencing each other. But the basic operation is just that easy. Don't make math, make music ;)

## The 'complex' mode

Hello, hello: switching to 'complex' does not mean making things 'complicated'. ;) But it is indeed a unique 'killer' feature that you might need to become familiar with.



This mode is available for both opto and vca designs. Generally speaking, the switch enables a second compressor stage after the other.

What?

Well, try serializing two compressors of your choice. You might come to the conclusion that each of the two does not 'work' as hard as a single instance. Instead thereof, you can go for less gain reduction in both of them and still end up with deep compression that most probably introduces less artifacts.

The 'complex' mode of BLOCKFISH takes the first instance as a gentle 'leveler'-type compressor with longer response times. The second instance has the job to 'catch' most of the transients that slipped through the first stage. The release phase is also shortened, so that you can drastically increase the overall loudness and make things 'compact' or 'dense'. Just like forming a BLOCK with your hands ;) )

With the complex mode turned on, the saturation stage is brought in-between the two stages. This can be helpful for achieving a very dense sound if the second stage is actually getting a saturated input signal: there may be just too little transients left for the second detector to measure; likewise, the average level is more taken into account.

## The controls

### Front plate view



*compression:*  
Amount of - guessed it! - compression



*response:*  
Turn left: fast attack, fast release. Turn right: make the thing react slower. In the opto mode, the 'release' time depends on the current average level and the 'memory' setting (->circuit view)



*output:*  
No need to say much here. Just note that the output stage won't cause any digital clipping since a final saturation stage takes care of limiting the audio to 0dBFS



*saturation:*  
Amount of additional harmonics through soft clipping either between compressor stage 1 and 2 (->'complex') or behind the single compressor. The 'timbre' of the saturation can be altered on the circuit board (see below)





*vca/opto:*  
Switches basic operation mode as described above

*low cut:*  
Removes low frequency content from the detector / opto circuit. Signal won't tend to pump like if switched off as most of the 'average' level in a signal usually resides at low frequencies

*air:*  
Takes the inverse gain reduction control voltage (-> some sort of expansion) and applies it to a high-passed portion of the input signal. Result is that the usual 'dampening' or 'muffling' effect of deep compression can be avoided. Used drastically, you can add 'gloss' to sources like vocals



*bypass:*  
Routes the audio in parallel to the entire circuit, thereby letting you compare the results to the unaffected signal and still monitor what's going on

*stereo:*  
Switches the main processing into stereo mode. However, detecting levels and applying gain reduction still takes place on both channels simultaneously.

### Circuit view (click on 'open')



*low cut freq:*  
Determines the cut-off frequency of the 'low cut' filter (->front plate view)

*air freq:*  
Sets the high-pass filter edge frequency where the inverse compression process is applied

*air level:*  
Simply the amount of air. Adjust to taste



*opto memory:*  
The gain control element consisting of a virtual light source and a photo resistor has a certain 'time lag' than can be adjusted here. Turned fully clockwise, the circuit has maximum memory, meaning that high signal levels will have some sort of 'charging' effect and elongate the release time. Turn anti-clockwise for fast, 'in-your-face' sounds

*sat emph freq:*  
This knob controls a low-shelf eq circuit that acts like a pre- and de-emphasis stage in conjunction with the saturation process. Its function is to filter out bass frequencies that would otherwise cause intermodulation distortion (which often sounds unpleasant). Turn it up to saturate mostly the mid and high region which makes the sound kind of 'thick' and 'warm'.



*dynamic sat:*  
Makes the circuit add saturation in a dynamic way by measuring the current transient information as well as the gain reduction. This is some sort of 'de-compression' process that adds a certain complexity to the resulting signal. Operation varies on the current mode: in the vca setting the circuit saturates symmetrically, adding only odd harmonics. In the opto mode, asymmetrical clipping is taking place, similar to a tube-driven circuit.

## SPITFISH de-esser



So you got a fine and clean digital studio. Congratulations. But sometimes, you feel like recording a real human. That's not a shame.

Well, ever been in this situation? You've recorded your (or any other's) golden voice and did everything right. The vocals were sweet, clear and fitted perfectly, soundwise, throughout the entire recording process.

But when it comes to mixing the song, you feel you have to go for a certain amount of compression. And there it happenssssss. Something sssspits right in your face, your ears as well as your speakers' tweeters are in deep trouble. The vocalist just sounds harsh, cold or slightly artificial. Then you've encountered that typical sibilance problem known as 'essing'.

The reasons can be many and the most common are:

- The singer is overdoing his/her performance in terms of 'ss' or 'shh' sounds
- The microphone is a condenser type, but has a peak in the upper midrange / high region (often called 'toppy' sound)
- The mic preamp is of poor quality and lacks openness and clarity with high-frequency content

Of course you can hire a better singer, rob a bank and get yourself some world-class equipment. But you should at least try a little de-essing.



Using such a dedicated processor can mean a dramatic impact on your recordings. If carefully applied, it might be able to turn mediocre sounds into great sounds by giving back a natural and homogenic performance.

The working principle of most de-essers is somehow similar, though the actual design can vary. Basically, the circuit is a specialized dynamics compressor that does the following:

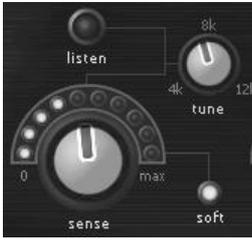
- Try to detect some sort of ssss-sounds. Most de-essers use a high-pass or band-pass filter tuned to the potential region of 'where it happens' (SPITFISH uses a band-band filter)
- Lower the level. Early designs did only reduce the whole signal by a certain amount of level that has been detected as 'essing'. Later units only filtered out the treble, often by a high-shelving filter. Better designs use a steep band-cut filter



digitalfishphones.com

## De-essing with SPITFISH

The SPITFISH detects 'essing' by being tuned to the most badly sounding region. To find out about where this is, press the 'listen' button.



You will probably hear... nothing.

This is okay, since you are listening to the signal in the same way the control circuit would get it.

In order to hear anything, you have to increase the 'sense' level until there is anything to sense at all.

The reason why you have to adjust something here is that you need to tell the device what's the most prominent level at where 'essing' is most likely to occur. Turning the

'sense' knob fully up means maximum sensitivity. You can compare this to a very low threshold on a classical compressor, though it's not quite the same.

Typically, you should adjust the level in a way that normal sounds don't let the meter instrument light up while 's'-like sounds trigger it as far as possible.

To achieve such maximum difference, unpress the 'soft' button. Now the unit is running in some sort of 'hyperactive' mode.



Now, the actual amount of de-essing can be adjusted via the 'depth' knob.

This controls the level of a band-cut filter tuned at the center frequency of where you set the detector at.

Whenever SPITFISH is behaving that sensitive as described above, you only need to dial in a small amount of 'depth' in order to achieve serious de-essing.

If it sounds too drastic, try using the 'soft' mode. This switch adjust the internal time constants to a slower behaviour and feeds the detector with a softer transition curve that rules over 'no' signal and 'fully essing'. Now, you can go for more 'depth' with a smoother behaviour, though it might sound as if SPITFISH is almost always taking away something.

But this can help greatly to even out an overly harsh signal.

Before I forget:

Of course, you are not restricted to vocals. You could as well try 'de-essing' cymbals or hi-hats on drum recording as well as smoothing out an entire recording. Just use your ears ;)

In order to be able to process stereo tracks, the unit also knows a 'stereo' mode. When pressed, the signal stays a stereo type but the detector takes a monaural sum of both channels to work with. This is normally not an issue.



digitalfishphones.com

## FLOORFISH expander / gate



If you are in a similar situation like me, having to deal with acoustic instruments from time to time, perhaps drum tracks, then you might have missed an expander on some occasions.

Maybe you also had the wish to lower the bleeding noise on your vocal track whenever the singer remains quiet. But cutting it completely off sounds way to cheesy.

Sadly, most audio applications or sequencers only include a basic noise gate which is pretty useless if you want to keep a signal's natural decay while lowering the noise floor.



I've designed the FLOORFISH to operate reliably on such noise floor (hence the name). The amount of attenuation can be controlled from 'don't do anything' to 'be a simple noise gate'.

### Quick FLOORFISHing

- Tune the device to the most prominent frequency in the source signal. You can use the listen button for monitoring purposes
- Select the basic operation mode. Similar to SPITFISH, the 'soft' mode also affects the softness of the transition curve (similar to a 'knee' control on some compression gear) as well as changing the attack & release times at low input levels. This prevents the unit from sounding 'choppy' around the noise floor level
- Adjust the sense level so that you see the LED beneath the knob light up when you want the signal to 'come through'. The LED represents the detector's behaviour. Likewise, the current attack & release times will affect the operation here
- Choose 'appropriate' time constants (attack & release). Generally, fast drum hits need almost always a 'zero attack' setting, depending on the 'tune' frequency. The 'release' should be as long as it would take the instrument to decay naturally. But you are free to use the device as an effect by setting up the times in a rhythmical manner
- Dial in the amount of expansion. Turning the big knob fully clockwise is equivalent to heavy gating. Being fully left means no expansion at all. The meter LEDs show the current amount of gain reduction

The 'bypass' and 'stereo' switches are doing the same as on the other two plugins. Thus, switching FLOORFISH to 'stereo' means a signal to remain stereo, but operation takes place on both channels in the same way.



digitalfishphones.com

**A final note:**

- Don't be fooled by the term 'analog-style'. This is not the analog world. Never ever. Even if we're aiming at it with great effort, true analog circuits still sound different and keep a certain magic. This is not a story of good or bad, it is the question of 'What do I want to achieve? What is my way of working? What is the weakest thing in my chain?'. We shouldn't forget that.
- Give me some feedback on any plugin of the fish fillets, of whatever kind. In the end, it all helps to improve this software and influence my future developments.
- Have fun.



Sascha Eversmeier  
Berlin, December 20, 2002  
<http://www.digitalfishphones.com>

*This program was written using Microsoft Visual C++ 5 and the Steinberg VST plugins software development kit (SDK).  
The Mac port was done by Urs Heckmann using CodeWarrior and ProjectBuilder.  
VST is a registered trade mark of Steinberg Media Technologies AG. All copyrights acknowledged.  
All other copyrighted trade marks belong to their respective owners.*



digitalfishphones.com

**PLEASE DO NOT HOST THIS SOFTWARE ON THE NET FOR YOURSELF. ALWAYS POINT A LINK TO MY PAGE (<http://www.digitalfishphones.com>) TO MAINTAIN UPDATED PROGRAM VERSIONS!!! Thank you ;)**

**LICENSE TERMS FOR THE 'FISH FILLETS' VST PLUGIN SOFTWARE (files: 'blockfish', 'spitfish', 'floorfish', in the following: 'software')**

The software is provided to the user 'as is'.

I make no warranties, either express or implied, with respect to the software and associated materials provided to the user, including but not limited to any warranty of fitness for a particular purpose. This software was purely written to meet my requirements, I do not warrant that the functions contained will meet yours, nor that the operation of it will be uninterrupted or error-free, or that defects in the software will be corrected.

I do not warrant or make any representations regarding the use or the results of the use of the software or any documentation provided therewith in terms of their correctness, accuracy, reliability, or otherwise. No information or advice given by me shall create a warranty or in any way increase the scope of this warranty.

I am not liable for any claims or damages whatsoever, including property damage, personal injury, intellectual property infringement, loss of profits, or interruption of business, or for any special, consequential or incidental damages, however caused.

At present this software is still under development and not to be a public release. It is freely given to people to test its functionality and allow feedback to me as the author for future development, though I do not warrant that there will be such.

The user is not allowed to distribute the program without my permission. Further, the user may not modify or may not decompile nor debug any the **fish fillets** software provided with this package.

**ANY PROGRAM IN THIS PACKAGE IS FREWARE.**

