Bomb Factory Plug-ins

Version 7.3

contents

chapter 1

Introduction

Welcome to the Bomb Factory™ plug-ins for Pro Tools|HD®, Pro Tools LE™, Pro Tools M-Powered™ and VENUE™ systems, brought to you by Digidesign®.

⚠ *References to Pro Tools LE™ in this guide are usually interchangeable with Pro Tools M-Powered™ or Pro Tools Academic™, except as noted in the Pro Tools M-Powered and Academic Getting Started Guides.*

Digidesign Bomb Factory plug-ins include the following:

Free Bomb Factory Plug-ins

- BF Essentials, which includes:
  - BF Essential Clip Remover
  - BF Essential Correlation Meter
  - BF Essential Meter Bridge
  - BF Essential Noise Meter
  - BF Essential Tuner
- Bomb Factory BF76
- Funk Logic Mastererizer

Other Bomb Factory Plug-ins

The following plug-ins are also available:

- Bomb Factory BF-3A
- Bomb Factory BF-2A
- Cosmonaut Voice
- Fairchild 660 and 670
- JOEMEEK VC5 Compressor
- JOEMEEK SC2 EQ
- Moogerfooger Analog Delay
- Moogerfooger Ring Modulator
- Moogerfooger 12-Stage Phaser
- Moogerfooger Lowpass Filter
- Pultec EQP-1A, EQH-2, and MEQ-5
- Purple Audio MC77
- SansAmp PSA-1
- Slightly Rude Compressor
- Tel-Ray Variable Delay
- Voce Spin
- Voce Chorus/Vibrato

Contents of the Boxed Version of Your Plug-in

Your plug-in package contains the following components:

- Installation disc
- Digidesign Registration Information Card
- One of the following authorization cards for authorizing plug-ins with an iLok USB Smart Key (not supplied):
  - Activation Card with an Activation Code

  – or –
  - License Card

System Requirements

To use Bomb Factory plug-ins you need the following:

- An iLok USB Smart Key
- An iLok.com account for managing iLok licenses
- One of the following:
  - A Digidesign-qualified Pro Tools system
  - A Digidesign-qualified Pro Tools system and a third-party software application that supports the Digidesign TDM, RTAS®, or AudioSuite plug-in standard
  - A qualified Avid® Xpress®, Avid Xpress DV or Avid DNA™ system
  - A Digidesign-qualified VENUE system (TDM only)

For complete system requirements, visit the Digidesign website (www.digidesign.com).

Compatibility Information

Digidesign can only assure compatibility and provide support for hardware and software it has tested and approved.

For a list of Digidesign-qualified computers, operating systems, hard drives, and third-party devices, visit the Digidesign website (www.digidesign.com).

Register Your Plug-ins

If you purchase a plug-in online from the Digi-Store (www.digidesign.com) using Internet Activation, you are automatically registered.

If you purchase a plug-in boxed version, check the included Digidesign Registration Information Card for information about how to register online.

Registered users receive periodic software update and upgrade notices.

Please refer to the Digidesign website (www.digidesign.com) or the Digidesign Registration Information Card for information on technical support.

Working with Bomb Factory Plug-ins

Refer to the *DigiRack Plug-ins Guide* for information on working with plug-ins, including:

- Inserting Plug-ins on Tracks
- The Plug-in Window
- Adjusting Parameters
- Automating Plug-ins
- Using the Librarian
- Clip Indicators

Conventions Used in This Guide

All Digidesign guides use the following conventions to indicate menu choices and key commands:

Convention	Action
File > Save	Choose Save from the File menu
Control+N	Hold down the Control key and press the N key
Control-click	Hold down the Control key and click the mouse button
Right-click	Click with the right mouse button

The following symbols are used to highlight important information:

☀ *User Tips are helpful hints for getting the most from your Pro Tools system.*

⚠ *Important Notices include information that could affect your Pro Tools session data or the performance of your Pro Tools system.*

⌨ *Shortcuts show you useful keyboard or mouse shortcuts.*

📖 *Cross References point to related sections in the Pro Tools Guides.*

About www.digidesign.com

The Digidesign website (www.digidesign.com) is your best online source for information to help you get the most out of your Pro Tools system. The following are just a few of the services and features available.

Registration Register your purchase online. See the enclosed registration form for details.

Support Contact Digidesign Technical Support or Customer Service; download software updates and the latest online manuals; browse the Compatibility documents for system requirements; search the online Answerbase or join the worldwide Pro Tools community on the Digidesign User Conference.

Training and Education Study on your own using courses available online or find out how you can learn in a classroom setting at a certified Pro Tools training center.

Products and Developers Learn about Digidesign products; download demo software or learn about our Development Partners and their plug-ins, applications, and hardware.

News and Events Get the latest news from Digidesign or sign up for a Pro Tools demo.

To learn more about these and other resources available from Digidesign, visit the Digidesign website (www.digidesign.com).

chapter 2

Installation

Installing Plug-ins

Installers for your plug-ins can be downloaded from the DigiStore (www.digidesign.com) or can be found on the plug-in installer disc (included with boxed versions of plug-ins). An installer may also be available on a Pro Tools software bundle installer disc.

Installation steps are essentially the same, regardless of the bundle, package, or system you purchase.

Updating Older Plug-ins

Because the Bomb Factory Plug-in installers contain the latest versions of Bomb Factory plug-ins, use them to update any plug-ins you already own.

⚠ *Be sure to use the most recent versions of Bomb Factory plug-ins available from the Digidesign website (www.digidesign.com).*

Installation

To install a plug-in:

1 Do one of the following:

- Download the installer for your computer platform from the Digidesign website (www.digidesign.com). After downloading, make sure the installer is uncompressed (.ZIP on Windows or .SIT on Mac).

– or –

- Insert the Installer disc into your computer.

2 Double-click the plug-in installer application.

3 Follow the on-screen instructions to complete the installation.

4 When installation is complete, click Finish (Windows) or Quit (Mac).

When you open Pro Tools, you are prompted to authorize your new plug-in.

Authorizing Bomb Factory Plug-ins

Bomb Factory plug-ins are authorized using the iLok USB Smart Key (iLok), manufactured by PACE Anti-Piracy, Inc.
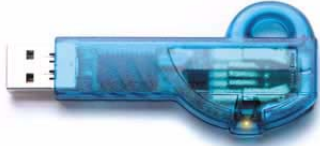
iLok USB Smart Key

☼ *Not all Bomb Factory plug-ins require authorization. For example, no authorization is required for Free Bomb Factory plug-ins.*

The iLok is similar to a dongle, but unlike a dongle, it is designed to securely authorize multiple software applications from a variety of software developers.

This key can hold over 100 licenses for all of your iLok-enabled software. Once an iLok is authorized for a given piece of software, you can use the iLok to authorize that software on any computer.

⚠ *The iLok USB Smart Key is not supplied with your plug-in or software option. You can use the one included with certain Pro Tools systems (such as Pro Tools|HD-series systems), or purchase one separately.*

If you purchased a plug-in online at the DigiStore (www.digidesign.com), authorize the plug-in by downloading licenses from iLok.com to an iLok.

📄 *See the iLok Usage Guide for details, or visit the iLok website (www.iLok.com).*

If you purchased a boxed version of a plug-in, it comes with either an Activation Code (on the included Activation Card) or an iLok License card:

- To authorize plug-ins using an Activation Code, see "Authorizing Plug-ins Using an Activation Code" on page 6.
- To authorize plug-ins using an iLok License Card, see "Authorizing Plug-ins Using a License Card" on page 7

Authorizing Plug-ins Using an Activation Code

To authorize a plug-in using an Activation Code:

1 If you do not have an existing iLok.com account, visit www.iLok.com and sign up for an iLok.com account.

2 Transfer the license for your plug-in to your iLok.com account by doing the following:

- Visit http://secure.digidesign.com/activation.
- Input your Activation Code (listed on your Activation Card) and your iLok.com User ID. Your iLok.com User ID is the name you create for your iLok.com account.

3 Transfer the licenses from your iLok.com account to your iLok USB Smart Key by doing the following:

- Insert the iLok into an available USB port on your computer.
- Go to www.iLok.com and log in.
- Follow the on-screen instructions for transferring your licences to your iLok.

📄 *For information about iLok technology and licenses, see the electronic PDF of the iLok Usage Guide.*

4 Launch Pro Tools.

5 If you have any installed unauthorized plug-ins or software options, you are prompted to authorize them. Follow the on-screen instructions to complete the authorization process.

Authorizing Plug-ins Using a License Card

License Cards are specific to each plug-in or software option. You will receive the appropriate License Cards for the plug-ins that you purchase. License Cards have a small punch-out plastic chip called a GSM cutout.

The authorization steps in this section must be repeated for purchased plug-in.

📖 *For additional information about iLok technology and authorizations, see the electronic PDF of the iLok Usage Guide.*

To authorize a plug-in using a License Card:

1 Insert the iLok into an available USB port on your computer.

2 Launch Pro Tools. You are prompted to authorize any installed unauthorized plug-ins or software options.

💡 *If you are already using a demo version of the plug-in or software option, launch Pro Tools before you insert the iLok, then insert the iLok into any available USB port when prompted by Pro Tools.*

3 Follow the on-screen instructions until you are prompted to insert the License Card into the iLok.

4 Separate the GSM cutout from the larger protective card by pulling it up and out with your thumb. Do not force the cutout down with your finger.

5 Insert the GSM cutout into the iLok. Visually verify that the metal portion of the cutout makes contact with the iLok's metal card reader.
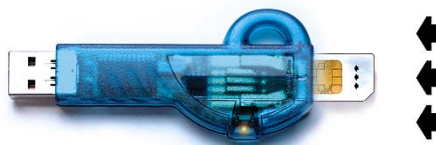
iLok with License Card

6 Follow the on-screen instructions to complete the authorization process for each plug-in.

7 After the authorization has completed, remove the GSM cutout from the iLok. (If you have to remove the iLok from the computer to remove the cutout, be sure to re-insert the iLok in any available USB port on your computer when you are finished.)

Removing Plug-ins

If you need to remove a plug-in from your system, follow the instructions below for your computer platform.

Windows

To remove a plug-in:

1 From the Start menu, choose Settings > Control Panel and double-click Add or Remove Programs.

2 Select the plug-in from the list of installed applications and click the Change/Remove button.

3 Follow the on-screen instructions to remove the plug-in.

4 When removal is complete, click OK to close the window.

Mac OS X

To remove a plug-in:

1 Locate and open the Plug-ins folder on your Startup drive (Library/Application Support /Digidesign/Plug-ins).

2 Do one of the following:

- Drag the plug-in to the Trash and empty the Trash.

– or –

- Drag the plug-in to the Plug-ins (Unused) folder.

chapter 3

Adjusting Bomb Factory Controls

Adjusting Plug-in Controls

You can adjust plug-in controls by dragging the control's slider or knob, or by typing a value into the control's text box. Additionally, some plug-ins have switches that can be enabled by clicking on them.

To adjust a plug-in control:

1 Begin audio playback so that you can hear the control changes in real time.

2 Adjust the controls of the plug-in for the effect you want. Refer to "Adjusting Controls Using a Mouse" on page 9 and "Editing Parameters Using a Computer Keyboard" on page 10.

Closing the plug-in will save the most recent changes.

Adjusting Controls Using a Mouse

You can adjust rotary controls by dragging horizontally or vertically. Parameter values increase as you drag upward or to the right, and decrease as you drag downward or to the left.

Keyboard Shortcuts

◆ For finer adjustments, Control-drag (Windows) or Command-drag (Mac) the control.

◆ To return a control to its default value, Alt-click (Windows) or Option-click (Mac) the control.

Editing Parameters Using a Computer Keyboard

Some controls have text boxes that display the current value of the parameter. You can edit the numeric value of a parameter with your computer keyboard.

If multiple Plug-in windows are open, Tab and keyboard entry remain focused on the plug-in that is the target window.

To change control values with a computer keyboard:

1 Click the text box corresponding to the control that you want to adjust.

2 Change the value.

- To increase a value, press the Up Arrow on your keyboard. To decrease a value, press the Down Arrow on your keyboard.

– or –

- Type the value.

💡 *In fields that support values in kilohertz, typing "k" after a number value will multiply the value by 1,000. For example, type "8k" to enter a value of 8,000.*

3 Do one of the following:

- Press Enter on the numeric keyboard to input the value and remain in keyboard editing mode.

– or –

- Press Enter on the alpha keyboard (Windows) or Return (Mac) to enter the value and leave keyboard editing mode.

💡 *To move forward through the different control fields, press the Tab key. To move backward, press Shift+Tab.*

Editing Parameters Using a Scroll Wheel

Some controls have text boxes that display the current value of the parameter. You can edit the numeric value of a parameter using a scroll wheel.

To change control values using a scroll wheel:

1 Click the text box corresponding to the control that you want to adjust.

2 To increase a value, scroll up with the scroll wheel. To decrease a value, scroll down with the scroll wheel.

Toggling Switches

To toggle a switch:

■ Click the switch on-screen.

BF Essentials Plug-ins

This chapter describes the following plug-ins in the BF Essential series:

- Clip Remover (see "BF Essential Clip Remover" on page 11)
- Correlation Meter (see "BF Essential Correlation Meter" on page 12)
- Meter Bridge (see "BF Essential Meter Bridge" on page 12)
- Noise Meter (see "BF Essential Noise Meter" on page 13)
- Tuner (see "BF Essential Tuner" on page 13)

TDM, RTAS, and AudioSuite support is noted for each plug-in.

BF Essential Clip Remover

(AudioSuite)

The BF Essential Clip Remover repairs clipped audio recordings. That red light no longer means a blown take! You'll be amazed how quickly this essential tool can repair clipped recordings. Best of all, it's much quicker and more accurate than using the Pencil tool. Set your levels very carefully. But when your signal gets too excited, try the BF Essential Clip Remover.

BF Essential Clip Remover

BF Essential Correlation Meter

(RTAS)

Solve tracking and mix problems, and trouble-shoot phase coherency with the BF Essential Correlation Meter. It works on stereo tracks or stereo submixes. Use it to stop phase problems before they start.

BF Essential Correlation Meter

BF Essential Meter Bridge

(RTAS)

The BF Essential Meter Bridge provides best-of-breed VU metering on any channel while using minimal DSP resources. Enjoy the ease of use afforded by a needle, a big meter, and a faithful emulation of the decades-old standard for meter ballistics. Select RMS or Peak metering, and calibrate instantly for useful viewing at any signal level, just like a pro tape machine.

BF Essential Meter Bridge

BF Essential Noise Meter

(RTAS)

The BF Essential Noise Meter is three meters in one:

- Set to "A," it's an A-weighted noise meter
- Set to "R-D," it's a Robinson-Dadson equal-loudness meter
- Set to "None," it's a VU meter with 100 DB of visual range.
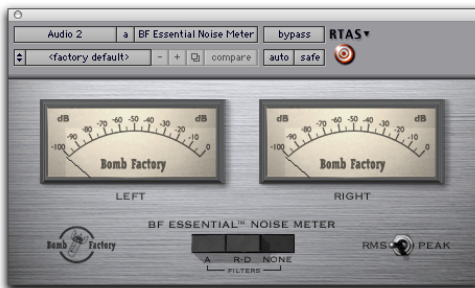
BF Essential Noise Meter

BF Essential Tuner

(RTAS)

Even though some quality plug-ins exist that give you access to a whole slew of guitar amps, cabinets, and effects, you still need to have a tuned instrument going into it. Why complicate things with another hardware device? Enter the BF Essential Tuner. In addition to tracking more accurately and tuning faster than any of those battery-eating handheld gizmos, it's more convenient than rackmount tuners.

BF Essential Tuner

Bomb Factory BF76

Bomb Factory BF76

(TDM, RTAS, and AudioSuite)

Modeled after the solid-state (transistor) 1176 studio compressor, the Bomb Factory BF76 preserves every sonic subtlety of this classic piece of studio gear.

How the Bomb Factory BF76 Works

The 1176 Compressor, originally introduced in the late 1970s, uses a FET (field-effect transistor). The 1176 also uses solid state amplification. The 1176 still provides an extremely high quality audio signal path, but because of these internal differences offers a much different compression sound than other compressors.

Four selectable compression ratios are provided, along with controls allowing variable attack and release times.
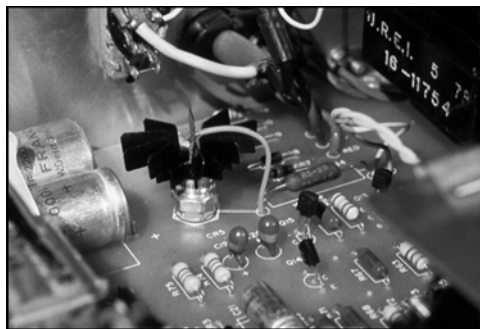
Deep inside the 1176

Various explanations overheard in the control room include "it's 100:1 compression ratio!" or equally adept quantitative analysis like "it makes it super squishy sounding." Now you can enjoy this sound—previously only available to super-serious-pro-engineers working in expensive pro recording studios—in the privacy of your own cubicle.

How the Bomb Factory BF76 is Used

The Bomb Factory BF76

The Input control sets the input signal level to the compressor, which, in the 1176 design, determines both the threshold and amount of peak reduction.

The Output control sets output level. Use it to bring the signal back to unity after applying gain reduction.

The Attack and Release controls set the attack and release times of the compressor. Full counterclockwise is slowest, and full clockwise is fastest. Attack times vary between 0.4 milliseconds to 5.7 milliseconds. Release times vary between 0.06 and 1.1 seconds.

The Ratio Push switches select the compression ratio from 4:1 to 20:1.

Finally, the Meter Push switches affect the metering.

- GR shows the amount of gain reduction.
- –18 and –24 show the output level (calibrated so that 0VU indicates –18dB FS and –24dB FS respectively).
- The "Off" switch turns off the meter.

Tips and Tricks

AudioSuite Processing

When using the AudioSuite version of the Bomb Factory BF76, be sure to select a side-chain input (normally the track you are processing). The default is "None" and if you leave it set like this, there's nothing feeding the detector and you won't hear any compression action.

Unexpected Visit from A&R Weevil Yields Instant Hit Mix

A favorite feature on one megabuck mixing console is its stereo bus compressor. With the flick of a switch, a punchy 8:1 compressor grabs the current mix producing "instant radio hit." Although Bomb Factory strongly disapproves of anything which adds further chaos to the already opaque A&R decision-making process, you may find that the Bomb Factory BF76 set to 8:1 ratio on a stereo mix provides a fast, easy alternate mix that can provide fresh ideas. It's also a handy way to make quick headphone submixes when tracking overdubs.

Give the Kids What They Want

Shift-click one of the Ratio Push switches to enable the "All Buttons In" mode. The compression ratio is still only 20:1, but the knee changes drastically and the compressor starts (mis)behaving a little bit like an expander—watch the meter for details. Hey, try it—sometimes it even sounds good.

Selecting Proper Attack and Release Times

As on the original unit, setting either the attack or release time too fast generates signal distortion. Again, this may or may not be the desired effect. A good starting point for attack and release is "6" and "3" (the defaults), and you can adjust as follows:

When compressing, use the slowest attack you can that preserves the desired dynamic range. Faster attacks remove the "punch" from the performance; slower attacks inhibit the compression you need to smooth things out.

When limiting, use the fastest attack time you can before you start to hear signal distortion in the low end. But don't sweat it too much: on the Bomb Factory BF76, the attack time ranges from "incredibly fast" to "really damn fast" by modern standards. It can be hard to hear the difference.

Release times are more critical on the Bomb Factory BF76. To set release times, listen for loud attacks and what happens immediately after the peaks. Set the release time fast enough that you don't hear unnatural dynamic changes, but slow enough that you don't hear unnecessary pumping between two loud passages in rapid succession.

Funk Logic Mastererizer

Funk Logic Mastererizer

(RTAS and AudioSuite)

Funk Logic Mastererizer is not an emulation of any particular piece of analog gear from the past. Instead, the Funk Logic Mastererizer is a sound design tool that adds some of the effects of gear in varying condition. It imparts that sometimes impossible-to-avoid sound of flawed equipment, media decay, and other low-fidelity benchmarks, and is designed for the creative degradation of audio.

Funk Logic Mastererizer

How the Mastererizer is Used

Insert the Funk Logic Mastererizer on a track. Adjust its controls as needed until the desired amount of hiss, hum, distortion, crackle, grit, dirt, or age is audible.

As always, experimentation provides the best results.

JOEMEEK Plug-ins

This chapter describes the JOEMEEK VC5 Meequalizer and the JOEMEEK SC2 Compressor RTAS and AudioSuite plug-ins.

JOEMEEK VC5 Meequalizer

(TDM, RTAS, and AudioSuite)

The JOEMEEK VC5 Meequalizer offers simple controls and incredibly warm, musical results.

How the Meequalizer Works

Picture this: drums in the spare bedroom. Microphones, cables, and recording gear strewn about the living room. A familiar scene—especially to legendary producer Joe Meek in 1962 as he prepared to record yet another chart topping hit.

Among countless other achievements, Joe Meek built custom gear to get the sounds in his head onto tape. One device was a treble and bass circuit with a sweepable mid control, built into a tiny tobacco tin! The Meequalizer VC5 recreates the exact circuitry used by Joe Meek.

JOEMEEK Meequalizer VC5 EQ

How the Meequalizer is Used

Operation of the Meequalizer is dead simple, and that's the whole point.

The Bass control adjusts low frequencies ±11.

The Mid and Mid Freq controls allow you to adjust mid frequencies, from 500Hz to 3.5KHz, ±11.

The Treble control adjusts high frequencies ±11.

Finally, the Gain control allows you to adjust the output level ±11.

(Yes, it goes up to 11.)

Tips and Tricks

Twelve O'Click

Alt-click (Windows) or Option-click (Mac) any knob to reset any knob to its unity position quickly.

JOEMEEK SC2 Compressor

(TDM, RTAS, and AudioSuite)

Add a little magic to your mixes with the JOEMEEK SC2 Compressor.

In use by top producers the world over, JOEMEEK compression is the secret weapon that gives your sound the character and excitement it deserves!

JOEMEEK SC2 Compressor

How the JOEMEEK Compressor Works

Legendary producer Joe Meek used to say, "If it sounds right, it is right." Nowhere is this more apparent than in Joe Meek's masterful use of non-linear, sometimes severe compression in his productions.

The JOEMEEK Compressor is designed purely as an effects compressor. Its purpose is to change the way the ear perceives sound; its action changes the clarity, balance and even rhythmic feel of music.

How the JOEMEEK Compressor is Used

Input Gain adjusts the input level to the compressor.

The Compression control affects the gain to the side-chain of the compressor. Use it along with Slope to adjust the amount of compression.

Output Gain provides makeup gain after compression.

Slope is similar to the compression ratio controls found on other compressors. However, on the JOEMEEK, the actual ratio varies based on program material so the term Slope is used instead. In practice, "1" is very gentle compression and "2" or "3" are typically right for voice and submixes. The higher numbers are better for instruments and extreme sounds. (At the suggestion of the original designers, Bomb Factory added the very cool "5" setting found on the later-model JOEMEEK SC2.2. Use "5" to create severe pumping effects.)

Attack sets the time that the compressor takes to act. Slower attacks are typically used when the sound of the compression needs to be less obvious.

Release sets the time during which signal returns to normal after compression. With longer release times, the compression is less noticeable.

Tips and Tricks

Not Perfect. Just Right.

Standard engineering practice says that a compressor should work logarithmically. For a certain increase of volume, the output volume should rise proportionally less, with a result that the more you put in, the more it's pushed down.

The JOEMEEK compressor doesn't work this way. As volume increases at the input, a point is reached where the compressor starts to work and the gain through the amplifier is reduced. If the input level keeps rising, gradually the gain reduction becomes less effective and the amplifier goes back to being a linear amplifier except with the volume turned down.

This is by design, and is based on an understanding of how the human ear behaves! The result is that the listener is fooled into thinking that the JOEMEEK compressed sound is louder than it really is—but without the strange psycho-acoustic effect of "deadness" that other compressors suffer from.

Overshoot

At fast Attack settings, it is possible to make the JOEMEEK "overshoot" on percussive program material. This means that the compression electronics are driven hard before the light cells respond to the increased level. The cells catch up and overcompress momentarily giving a tiny dip immediately following the start of the note.

To hear it, use a drum track, set Slope to "5," and Attack and Release to fastest. Used sparingly, this effect can contribute to musical drive in your tracks.

Attack/Release Times

It may be difficult to understand the interactions between the Attack and Release controls, because the JOEMEEK Compressor behaves very differently than typical compressors. Experimentation is the best option, but an explanation may help you understand what's going on.

The JOEMEEK Compressor uses a compound release circuit that reacts quickly to short bursts of volume, and less quickly to sustained volume. While the unit was being prototyped and designed, the values and ranges of these timings were chosen by experimentation using wide ranges of program material.

Because of these intentional effects produced by the compressor, the JOEMEEK makes a perfect tool for general enhancement of tracks to "brighten," "tighten," "clarify" and catch the attention of the listener, functions that are difficult or impossible to achieve with conventional compressor designs.

chapter 8

Bomb Factory BF-3A

Bomb Factory BF-3A

(TDM, RTAS, and AudioSuite)

Bomb Factory extends its award-winning Classic Compressors line with the BF-3A, based on the classic LA-3A. A secret weapon of pros in the know, the LA-3A adds a smoothness and sonic texture that makes sounds jump right out of the mix.

How BF-3A Works

Designed and manufactured in the late 1960s, the original LA-3A shares many components in common with the LA-2A compressor. Just like the LA-2A, the heart of the LA-3A is the T4B Electro-Optical Attenuator. This is a device that converts audio to light and back and is largely responsible for the compression character of the unit.

BF-3A

While the LA-2A's gain comes from a tube amplifier, the LA-3A's gain comes from a solid-state (transistor) amplifier. This gives the LA-3A a solid midrange and more aggressive tone. Other subtle modifications change the behavior of the T4B, causing it to respond differently—particularly in response to percussive material.

How BF-3A is Used

The LA-3A is famous for its unique sonic imprint on guitar, piano, vocals and drums. Because it's so easy to control, you'll be getting classic tones in no time with the BF-3A.

Setting the Compression Action

The Peak Reduction and Output Gain controls combine with the Comp/Limit switch to determine the amount and sound of the compression.

Peak Reduction controls the amount of signal entering the side-chain. The more Peak Reduction you dial in, the more "squashed" and compressed the sound will be. Too little peak reduction and you won't hear any compression action; too much and the sound becomes muffled and dead sounding.

Output Gain provides makeup gain to make the signal louder after passing through the peak reduction.

The Comp/Limit switch affects the compression ratio. The common setting for audio production is Comp, which provides a maximum compression ratio of approximately 3:1. In Limit mode, the unit behaves more like a broadcast limiter, with a higher threshold and compression ratio of approximately 15:1.

Metering

Both Gain Reduction and Output metering are provided.

In GR (Gain Reduction) mode, the needle moves backward from 0 to show the amount of compression being applied to the signal in dB.

In Out (Output) mode, the needle indicates the output level of the signal. The meter is calibrated with 0 VU indicating –18dBFS.

Tips and Tricks

AudioSuite Processing

When using the AudioSuite version of the BF-3A, be sure to select an auxillary side-chain input (normally the track you are processing). The default is "None" and if you leave it set like this, there is nothing feeding the detector and you will not hear any compression action.

Line Amp

Turn the Peak Reduction knob full counter-clockwise (off) and use the Gain control to increase the signal level. Although the BF-3A does not compress the sound with these settings, it still adds its unique character to the tone.

chapter 9

Bomb Factory BF-2A

Bomb Factory BF-2A

(TDM, RTAS, and AudioSuite)

Meticulously crafted to capture every nuance of the legendary LA-2A tube studio compressor, the Bomb Factory BF-2A provides the most authentic vintage compression sound available.

How BF-2A Works

Designed and manufactured in the early 1960s, the LA-2A achieved wide acclaim for its smooth compression action and extremely high quality audio signal path.

Originally designed as a limiter for broadcast audio, a Comp/Limit switch was added to LA-2A compressors after serial number 572. The subsequent addition of a Comp (Compress) setting made the LA-2A even more popular for use in audio production. However, the switch was inconveniently located on the back of the unit next to the terminal strips and tube sockets in the original version. In the BF-2A plug-in, the switch has been placed on the front panel, where you can make better use of it.

BF-2A

The heart of the LA-2A is its patented T4B ElectroOptical Attenuator, which provides the compression action. The T4B consists of a photo-conductive cell, which changes resistance when light strikes it. It is attached to an electro-luminescent panel, which produces light in response to voltage. Audio (voltage) is applied to the light source, and what happens as the audio converts to light and back to voltage gives the LA-2A its unique compression action. (Yes, the Bomb Factory BF-2A preserves all the subtle characteristics of this unique electronic circuit.) After compression, gain brings the signal back to its original level. The LA-2A's gain comes from a tube amplifier, which imparts further character to the tone. In fact, it's common to see engineers using the LA-2A simply as a line amp, without any compression applied to the signal.

How BF-2A is Used

One beautiful side effect of the LA-2A's elegant design is that it's easy to hear the compression action. When the BF-2A's two knobs are set properly, you know you got it right. It's a great unit for learning the art of compression!

Setting the Compression Action

The Peak Reduction and Gain controls combine with the Comp/Limit switch to determine the amount and sound of the compression.

Peak Reduction controls the amount of signal entering the side-chain, which in turn affects the amount of compression and the threshold. The more Peak Reduction you dial in, the more "squashed" the sound. Too little peak reduction and you will not hear any compression action; too much and the sound becomes muffled and dead sounding.

Gain provides makeup gain to bring the signal back after passing through peak reduction.

The Comp/Limit switch affects the compression ratio. The common setting for audio production is Comp, which provides a maximum compression ratio of approximately 3:1. In Limit mode, the unit behaves more like a broadcast limiter, with a higher threshold and compression ratio of approximately 12:1.

Using the Side-Chain Filter

The BF-2A provides an extra parameter, a side-chain filter, that does not have a control on the plug-in interface, but that can be accessed on-screen through Pro Tools automation controls. In addition, the side-chain filter can be adjusted directly from any supported control surface.

This side-chain filter reproduces the effect of an adjustable resistor on the back panel of the LA-2A. This control cuts the low frequencies from the side-chain, or control signal, that determines the amount of gain reduction applied by the compressor.

By increasing the value of the side-chain filter, you filter out frequencies below 250 Hz from the control signal, and decrease their effect on gain reduction.

◆ A setting of zero means that the filter is not applied to the side chain signal.

◆ A setting of 100 means that all frequencies below 250 Hz are filtered out of the side chain signal.

To access the side-chain filter on-screen:

1 Click the Auto button in the BF-2A plug-in window to open its Automation Enable window.

2 In the list of parameters at the left, click to select Side-Chain Filter and click Add (or, simply double-click the desired parameter in the list).
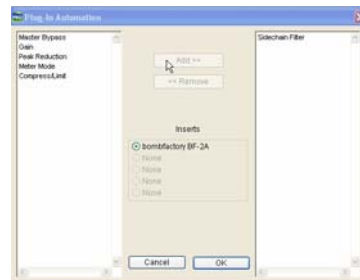
Enabling a parameter in the Plug-in Automation window

3 Click OK to close the plug-in automation window.

4 Click the Track Display selector and choose Side-Chain Filter from the BF-2A sub-menu.

Displaying an automation-enabled plug-in parameter

5 Edit the breakpoint automation for the BF-2A side-chain filter. Control range is from 0 (the default setting where no filtering is applied to the side-chain) to 100% (maximum side-chain filtering).

To access the side-chain filter from a control surface:

1 Focus the BF-2A plug-in on your control surface.

2 Adjust the encoder or fader current targeting the Side-Chain Filter parameter.

To automate your adjustments, be sure to enable automation for that parameter as described above. See the Pro Tools Reference Guide for complete track automation instructions.

Metering

Both Gain Reduction and Output metering are provided.

In Gain Reduction mode, the needle moves backward from 0 to show the amount of compression being applied to the signal in dB.

In Output mode, the needle indicates the output level of the signal. The meter is calibrated with 0 VU indicating –18 dBFS.

Tips and Tricks

AudioSuite Processing

When using the AudioSuite version of the BF-2A, be sure to select an auxillary side-chain input (normally the track you're processing). The default is "None" and if you leave it set like this, there is nothing feeding the detector and you will not hear any compression action.

Line Amp

Turn the Peak Reduction knob full counter-clockwise (off) and use the Gain control to increase the signal level. Although the BF-2A does not compress the sound with these settings, it still adds its unique character to the tone.

Feed the BF-2A into the BF76

Or vice versa. Glynn Johns (who has worked with the Stones, the Who, and others) popularized the early '70s British trick of combining a slower compressor with a faster one. The effect can produce very interesting sounds! Try applying Peak Reduction using the BF-2A, then squash the missed attacks using the faster BF76.

Purple Audio MC77

Purple Audio MC77

(TDM, RTAS, and AudioSuite)

The Purple Audio MC77 is a spot-on digital replica of Andrew Roberts's acclaimed MC77 Limiting amplifier, which in turn is an update of his classic MC76 hardware unit. Representing a different take on the 1176-style FET limiter, the Bomb Factory Purple Audio MC77 preserves every audio nuance and sonic subtlety of the classic originals.

How the Bomb Factory Purple Audio MC77 Works

The Bomb Factory MC77

The Bomb Factory Purple Audio MC77 has controls identical in name to those of the Bomb Factory BF76, and which function similarly. See Chapter 5, "Bomb Factory BF76" for more information.

Tel-Ray Variable Delay

Tel-Ray Variable Delay

(TDM, RTAS, and AudioSuite)

Ad-N-Echo to any voice or instrument using the Tel-Ray Variable Delay. It provides lush delay, amazing echo, and warms up your tracks and mixes.
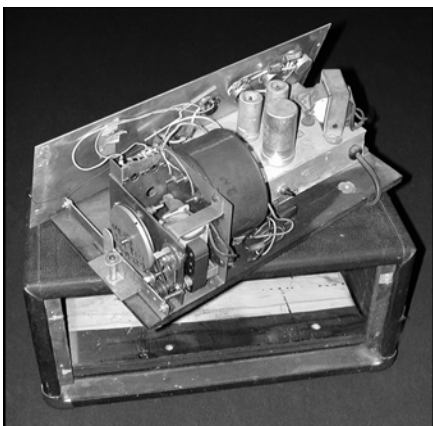
Space-age technology in a can

How the Tel-Ray Works

In the early 1960s, a small company experimented with electronics and technology. When they came up with something great, they would Tell Ray (the boss).

One invention involved a tuna can, a motor, and a few tablespoons of cancer-causing oil. The creation: an Electronic Memory Unit. A technology, they were sure, that would be of great interest to companies like IBM and NASA.

Though it never made it to the moon, most every major guitar amp manufacturer licensed the killer technology that gives Tel-Ray its unique sound.

How the Tel-Ray is Used

Input/Output Section

Input sets the signal level to the tuna can echo unit.

Tel-Ray Variable Delay

Tone is a standard tone control like those commonly found on guitar effects.

Mix adjusts the amount of dry (unprocessed) signal relative to the amount of wet (processed) signal. Full clockwise is 100% wet. (On original units, this control is located deep inside the box, typically soaked in carcinogenic PCB oil.)

Output is a simple digital output trim control.

Echo/Delay Section

Variable Delay selects the delay time. Delay times vary from 0.06 to 0.3 seconds. Full clockwise is slowest.

Variation adjusts how much variation occurs in the delay. The more variation you use, the more warbled and wobbly the sound becomes.

Sustain determines how long the delay takes to die out. It is actually a feedback control similar to the one found on the Moogerfooger Analog Delay.

Echo/Doubler determines whether or not a second record head is engaged, resulting in a double echo.

Tips and Tricks

Variation? Do They Ever.

Each and every Tel-Ray we tested (and Bomb Factory owns more than a dozen) varied drastically in motor and flywheel stability, resulting in different pitch and variation effects. The same unit even sounded different day to day, depending on temperature, warm-up time and other factors.

Since the original units are basically thirty year-old tuna cans bolted to plywood with springs and motors flopping around inside, Bomb Factory added the Variation knob so you can dial in a Tel-Ray in whatever state of disrepair you desire.

Moogerfooger Plug-ins

This chapter describes the following Mooger-fooger plug-ins:

- Moogerfooger Analog Delay (see "Moogerfooger Analog Delay " on page 35)
- Moogerfooger Ring Modulator (see "Moogerfooger Ring Modulator" on page 37)
- Moogerfooger 12-Stage Phaser (see "Moogerfooger 12-Stage Phaser " on page 39)
- Moogerfooger Low-Pass Filter (see "Moogerfooger Low-Pass Filter" on page 41)

Moogerfooger Analog Delay

(TDM, RTAS, and AudioSuite)

The Moogerfooger Analog Delay is simply the coolest (and warmest sounding!) delay in the digital domain.

Moogerfooger Analog Delay

How the Moogerfooger Analog Delay Works

A Delay Circuit produces a replica of an audio signal a short time after the original signal. Mixed together, the delayed signal sounds like an echo of the original. And if this mixture is fed back to the input of the delay circuit, the delayed output provides a string of echoes that repeat and die out gradually. It's a classic musical effect.

The Moogerfooger Analog Delay uses Bucket Brigade Analog Delay Chips to achieve its delay. These analog integrated circuits function by passing the audio waveform down a chain of thousands of circuit cells, just like water being passed by a bucket brigade to put out a fire. Each cell in the chip introduces a tiny time delay. The total time delay depends on the number of cells and on how fast the waveform is "clocked," or moved from one cell to the next.

With the advent of digital technology, these and similar analog delay chips have gradually been phased out of production. In fact, Bob Moog secured a supply of the last analog delay chips ever made, and used them to build a Limited Edition of 1,000 "real-world" Moogerfooger Analog Delay units.

So Why Analog?

Compared to digital delays, the frequency and overload contours of well-designed analog delay devices generally provide smoother, more natural series of echoes than digital delay units. Another difference is that the echoes of a digital delay are static because they are the same digital sound repeated over and over, whereas a bucket brigade device itself imparts a warm, organically evolving timbre to the echoes.

Of course, Bomb Factory's digital replica re-creates all the warm, natural sounds of its analog counterpart.

Not Better—Different

Working directly with Bob Moog, Bomb Factory enhanced the Moogerfooger Analog Delay to be even more useful for digital recording. An integrated Highpass Filter allows you to remove unwanted bass buildup from the feedback loop, allowing you to have warmer, more-controllable echo swarms while minimizing the potential for digital clipping.

How the Moogerfooger Analog Delay is Used

Delay Section

Delay Time allows you to select the length of delay between the original and the delayed signal. Used with Feedback, it also affects how long apart the echoes are.

The Short/Long switch sets the range of the Delay Time control. Set to Short, the Delay Time ranges from 0.04 to 0.4 seconds. Set to Long, it ranges from 0.08 to 0.8 seconds.

Feedback determines how much signal is fed back to the delay input, affecting how fast the echoes die out.

Filter Section

The Highpass knob removes low frequencies from the feedback loop. It removes undesirable low frequency "mud" common when mixing with delays and also allows the creation of amazing echo swarms that won't clip the output. Dial in a highpass frequency from 50 Hz to 500 Hz. Frequencies below the setting are filtered from the feedback loop.

The HPF Off/HPF On enables or disables the highpass filter (HPF).

Audio Input and Mixing

The remaining controls set the input level and effect mix.

The Drive control sets the input gain.

The Mix control blends the original input signal with the delayed signal.

LED Indicators

Three LEDs down the center of the unit provide visual feedback. Level glows green when signal is present. HPF turns green when the highpass filter is enabled. And Bypass glows either red (bypassed) or green (not bypassed) to show whether or not the effect is in the signal path.

Tips and Tricks

Infidelity

Because analog delay chips offer only a fixed number of cells, the extended delay times store a lower-fidelity version of the input signal. Try the Long delay setting when going for cool "lo-fi" sounds and textures.

Echo Swarms

By carefully adjusting the Feedback, Drive, and Highpass controls, you can use the Mooger-fooger Analog Delay as a sound generator. Simply pulse the delay unit with a short piece of audio (even a second will do), and adjust the Delay Time knob. Set correctly, the unit will generate cool timbres for hours all by itself.

Moogerfooger Ring Modulator

(TDM, RTAS, and AudioSuite)

The Moogerfooger Ring Modulator provides a wide-range carrier oscillator and dual sine/square waveform LFO. Add motion to rhythm tracks and achieve radical lo-fidelity textures—you set the limits!

How the Moogerfooger Ring Modulator Works

Like the Lowpass Filter, the Moogerfooger Ring Modulator has its roots in the original MOOG Modular synthesizers. It provides three classic MOOG modules: a Low Frequency Oscillator, a Carrier Oscillator, and a Ring Modulator.

Low Frequency Oscillators (or LFOs) create slow modulations like vibrato and tremolo. The LFO in the Moogerfooger Ring Modulator is a wide-range, dual-waveform (sine/square) oscillator.

Moogerfooger Ring Modulator

The Carrier Oscillator is a wide-range sinusoidal oscillator. It's called the Carrier Oscillator because, like the carrier of an AM radio signal, it's always there, ready to be modulated by the input.

A Ring Modulator takes two inputs, and outputs the sum and difference frequencies of the two inputs. For example, if the first input contains a 500 Hz sine wave, and the second input contains a 100 Hz sine wave, then the output contains a 600 Hz sine wave (500 plus 100) and a 400 Hz (500 minus 100) sine wave.

How the Moogerfooger Ring Modulator is Used

LFO Section

Control the LFO using the Amount and Rate knobs and the Square/Sine waveform selector switch.

Amount determines the amount of LFO waveform that modulates the frequency of the carrier oscillator. When the knob is full counterclockwise, the carrier is unmodulated. Fully clockwise, the carrier oscillator is modulated over a range of three octaves.

Rate determines how fast the LFO oscillates, from 0.1 Hz (one cycle every ten seconds) to 25 Hz (twenty-five cycles per second). The LFO light blinks to give a visual indication of the LFO rate.

The Square/Sine switch selects either a square or sine waveform. The square wave produces trill effects, whereas the sine waveform produces vibrato and siren effects.

Modulator Section

The Carrier Oscillator is controlled by the Frequency knob and the Low/High switch.

In the Low position, the Frequency knob ranges from 0.5 Hz to 80 Hz. In the High position, the Frequency knob ranges from 30 Hz to 4 kHz. Operating at the selected frequency, the carrier oscillator provides one input to the ring modulator, with the other coming from the input signal.

Audio Input and Mixing

The Drive control sets the input gain.

The Mix control blends the input signal and the Ring Modulator output. You hear only the input signal when the knob is counterclockwise, and only the ring modulated signal with the knob fully clockwise

LED Indicators

Three LEDs provide visual feedback. Level glows green when signal is present. LFO blinks to show the LFO rate. And Bypass glows either red (bypassed) or green (not bypassed) to show whether or not the effect is in the signal path.

Tips and Tricks

A Little Goes a Long Way

You'll discover tons of great uses for the Moogerfooger Ring Modulator through experimentation. But don't forget to try using it in subtle ways, adding "just a hint" to harshen up or add a metallic quality to individual tracks buried in the mix. Almost all the great MOOG sounds feature subtle, clever uses of Ring Modulation.

Moogerfooger 12-Stage Phaser

(TDM, RTAS, and AudioSuite)

The Moogerfooger 12-Stage Phaser combines a 6- or 12-stage phaser with a wide-ranging variable LFO. Start with subtle tremolo or radical modulation effects, then crank the distortion and resonant filters for unbelievable new tones—all featuring classic MOOG® sound.

How the Moogerfooger 12-Stage Phaser Works

The Moogerfooger 12-Stage Phaser offers 6 or 12 stages of MOOG resonant analog filters. Unlike the Lowpass Filter, however, the filters are arranged in an allpass configuration.
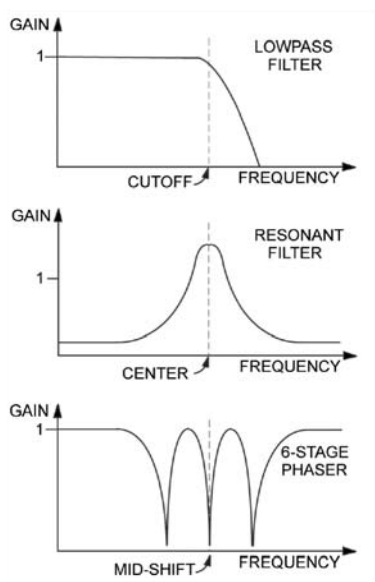
Different types of filters

A phaser works by sweeping the mid-shift frequency of the filters back and forth. As this happens, the entire frequency response of the output moves back and forth as well. The result is the classic phaser "whooshing" sound as different frequency bands of the signal are alternately emphasized and then attenuated.

A sweep control allows you to adjust the range of the frequency shift. And, keeping in the spirit of the MOOG modular synthesizers, an integrated LFO allows you to modulate the sweep control, allowing for extreme effects.

How the Moogerfooger 12-Stage Phaser is Used

Moogerfooger 12-stage Phaser

LFO Section

Control the LFO using the Amount and Rate knobs and the Lo/Hi selector switch.

Amount varies the depth of phaser modulation, from barely perceptible at the full counterclockwise position, to the full sweep range of the phaser (full clockwise or "Kill" setting).

Rate determines how fast the LFO oscillates. The LFO light blinks to give a visual indication of the LFO rate.

The Lo/Hi switch selects the range of the Rate control. When the switch is Lo, the Rate control varies from 0.01 Hz (one cycle every hundred seconds) to 2.5 Hz (2.5 cycles every second). When the switch is Hi, the Rate control varies from 2.5 Hz (2.5 cycles every second) to 250 Hz (two hundred fifty cycles per second). With such a wide range of rates available, obviously you'll need to adjust Rate after you flick the Lo/Hi switch to get the sound you desire.

Phaser Section

Control the Phaser with the Sweep and Resonance knobs and the 6-Stage/12-Stage switch.

Resonance adjusts the feedback of the analog filters. As you add more resonance, the peaks caused by the filters get sharper and more noticeable.
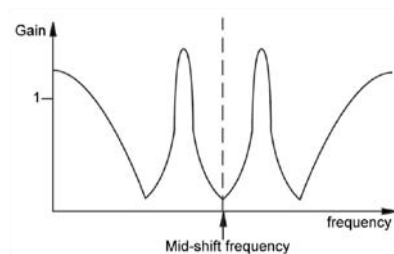
Responses of a phaser with high resonance

Sweep adjusts the center frequency point of the filters. Use it in conjunction with Amount to control the frequencies affected by the phaser.
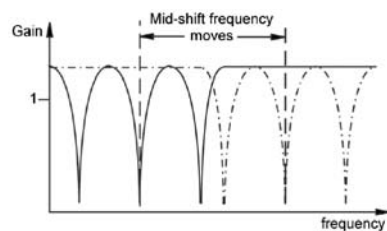
Sweep adjusts the center frequency point

Audio Input

Finally, the Drive control sets the input gain.

LED Indicators

Three LEDs provide visual feedback. Level glows green when signal is present. LFO blinks to show the LFO rate. And Bypass glows either red (bypassed) or green (not bypassed) to show whether or not the effect is in the signal path.

Tips and Tricks

More Harmonics = More Fun

The richer the harmonic content of the sound, the more there is to filter and sweep. Try adding distortion using the SansAmp PSA-1 before the phaser–it's a cool variation on the common signal path used when putting a phaser in front of a guitar amp.

Aggressive. Extreme.

Dr. Moog apparently took these mantras of early 21st Century recording science to heart when he designed the Rate knob on his phaser. Flick the Rate switch to Hi and let the party begin. Try muting a track and mixing in bits of extremely phase-swept material.

It's An Effect—Play With It

All the controls on the Moogerfooger 12-Stage Phaser are fully independent of one another. This means you can set them in any combination that you wish. There is no such thing as a "wrong" combination of settings, so you can experiment all you like to find new, exciting effects for your music.

Moogerfooger Low-Pass Filter

(TDM, RTAS, and AudioSuite)

The Moogerfooger Lowpass Filter features a 2-pole/4-pole variable resonance filter with envelope follower. Use it to achieve classic 60s and 70s sounds on bass and electric guitar, or just dial in some warm, fat analog resonance when you need it.
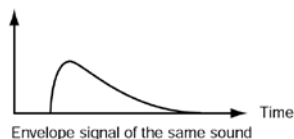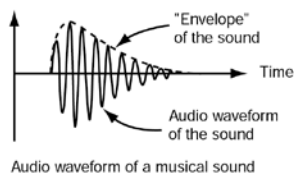
How the Moogerfooger Lowpass Filter Works

With the invention of the MOOG® synthesizer in the 1960s, Bob Moog started the electronic music revolution. A direct descendent of the original MOOG Modular synthesizers, the Moogerfooger Lowpass Filter provides two classic MOOG modules: a Lowpass Filter and an Envelope Follower.

A Lowpass Filter allows all frequencies up to a certain frequency to pass, and cuts frequencies above the cutoff frequency. It removes the high frequencies from a tone, making it sound more mellow or muted. The Moogerfooger Lowpass Filter contains a genuine four-pole lowpass filter. We say "genuine" because the four-pole filter—a major part of the "MOOG Sound" of the 60s and 70s—was first patented by Bob Moog in 1968! Bob worked directly with Bomb Factory to ensure that the digital version preserved all the character, nuances, and personality of his original classic analog design.

An Envelope Follower tracks the loudness contour, or envelope, of a sound. Think of it like this: each time you play a note, the envelope goes up and then down. The louder and harder you play, the higher the envelope goes. In the Moogerfooger Lowpass Filter, the Envelope Follower drives the cutoff frequency of the Lowpass Filter. Since the envelope follows the dynamics of the input, it "plays" the filter by sweeping it up and down in response to the loudness of the input signal.

Envelope

How the Moogerfooger Lowpass Filter is Used

*Moogerfooger Low-Pass Filter*

Envelope Section

The Amount knob determines how much the envelope varies the filter. When the knob is counterclockwise, the envelope signal has no effect on the filter. When the knob is fully clockwise, the envelope signal opens and closes the filter over a range of five octaves.

The Smooth/Fast switch determines how closely the envelope tracks the loudness of the input signal. Some sounds (like guitar chords) have long, rough envelopes, and often sound better with less dramatic changes in the filter. Other sounds (like bass or snare drum) are quick and sharp, and sound great when the filter closely tracks their attack.

Filter Section

Control the filter using the Cutoff and Resonance knobs and the 2-Pole/4-Pole switch.

Cutoff opens and closes the filter. Turned counterclockwise, fewer high frequencies pass through the filter. Turned clockwise, more high frequencies pass.
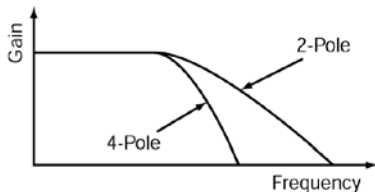
The 2/4 pole switch selects the filter slope

Resonance changes the way the filter sounds. At low resonance, low frequencies come through evenly. At high resonance, frequencies near the cutoff frequency are boosted, creating a whistling or vowel-type quality. When resonance is maxed out, the filter oscillates and produces its own tone at the cutoff frequency. This oscillation interacts with other tones as they go through the filter, producing the signature Moog sound.

The 2-Pole/4-Pole switch selects whether the signal goes through half the filter (2-pole) or the entire filter (4-pole). 2-pole is brighter, while 4-pole has a deeper, mellow quality.

Audio Input and Mixing

The remaining controls set the input level and effect mix.

The Drive control sets the input gain. Use it to adjust the input to the filter and envelope follower for desired impact.

The Mix control blends the original input signal with the filtered signal. Use it to get any mixture of filtered and unfiltered sound.

LED Indicators

Three LEDs down the center of the unit provide visual feedback. Level glows green when signal is present to the envelope circuit. Env (envelope) glows redder in response to the envelope tracking of the input. And Bypass glows either red (bypassed) or green (not bypassed) to show whether or not the effect is in the signal path.

Tips and Tricks

Auto Wah Using an External LFO

Try inserting an LFO ahead of the Moogerfooger Lowpass Filter to produce a cool "auto wah." Or use Voce Spin's rotating speaker for even trippier sounds!

chapter 13

SansAmp PSA-1

SansAmp PSA-1

(TDM, RTAS, and AudioSuite)

Punch up existing tracks or record great guitar sounds with the SansAmp PSA-1. Capture bass or electric guitar free of muddy sound degradation and dial in the widest range of amplifier, harmonic generation, cabinet simulation and equalization tone shaping options available!

How the PSA-1 Works

B. Andrew Barta of Tech 21, Inc. introduced the SansAmp Classic in 1989. A guitar player with both a trained ear and electronics expertise, Andrew and Tech 21 pioneered the market for tube amplifier emulation.

SansAmp's FET-hybrid circuitry captures the low-order harmonics and sweet overdrive unique to tube amplifiers. And pushed harder, SansAmp also generates cool lo-fi and grainy sound textures that still retain warmth.

SansAmp also features a proprietary speaker simulator which emulates the smooth, even response of a multiple-miked speaker cabinet—free of the harsh peaks, valleys and notches associated with single miking or poor microphone placement.

Finally, SansAmp provides two extremely sweet sounding tone controls (high and low) that sound great on most anything.

Tube sound, speaker simulation, warm equalization and cool lo-fi textures—no wonder thousands of records feature the classic sounds of SansAmp!

How the PSA-1 is Used

Use the eight knobs to dial in your desired tone or effect.

SansAmp

Pre-Amp

Pre-Amp determines the input sensitivity and pre-amp distortion. Increasing the setting produces an effect similar to putting a clean booster pedal ahead of a tube amp, overdriving the first stage. For cleaner sounds, use settings below the unity-gain point.

Buzz

Buzz controls low frequency break up and overdrive. Boost the effect by turning clockwise from the center point indicated by the arrows. As you increase towards maximum, the sound becomes (you guessed it) buzzy, with added harmonic content. For increased clarity and definition when using distortion, position the knob at its midpoint or towards minimum.

Punch

Punch sets midrange break up and overdrive. Decreasing from the center produces a softer, "Fender"-style break up. Increasing the setting produces a harder, heavier distortion. At maximum, it produces a sound similar to a wah pedal at mid-boost position placed in front of a Marshall amp.

Crunch

Crunch brings out upper harmonic content and, on guitars, pick attack. For cleaner sounds or smoother high end, decrease as needed.

Drive

Drive increases the amount of power amp distortion. Power amp distortion is associated with the "Vintage Marshall" sound—using SansAmp, you can produce the effect even at low levels.

Low

Low provides a tone control specially tuned for maximum musicality when used to EQ low frequencies on instruments. Boost or cut by ±12 dB by turning from the center point indicated by the arrows.

High

Use High to boost or cut high frequencies ±12 dB.

Level

Use Level to re-establish unity after adding distortion or equalizing the signal.

Tips and Tricks

Peace and Unity

A little known fact: The arrows in the SansAmp controls indicate the unity-gain position.

Louder and Cleaner

For best results, don't set the Pre-Amp level lower than unity gain when the Drive knob is at 9 o'clock or higher. However, if you want a crystal-clear sound and the Drive control is already near minimum, decrease Pre-Amp to further remove distortion.

Pre-Amp Versus Drive

To create varying types of overdrive, vary Pre-Amp in relation to Drive. A high Pre-Amp setting emphasizes pre-amp distortion (see "Mark 1" preset), while high Drive settings emphasize power amp distortion (see "Plexi" preset).

Voce Plug-ins

This chapter describes the Voce™ Chorus/Vibrato and Voce Spin plug-ins.

Voce Chorus/Vibrato

(TDM, RTAS, and AudioSuite)

Voce Chorus/Vibrato recreates the mechanical scanner vibrato found in the B-3 Organ. Three settings of chorus and three settings of vibrato presented on one cool knob! Fun and easy to use, it's a classic effect used for over sixty years.

How Voce Chorus/Vibrato Works

In a large pipe organ, "ranks" of pipes (multiple pipes designed to emit the same frequency) aren't perfectly in tune. The effect goes by the name "multirank" or, more commonly, "chorus."

Inside every B-3 organ, on the end of the drive-shaft that spins the tonewheels, you'll find a mechanical contraption that delays the sound of the organ. Originally added to make the B-3 sound more like a pipe organ, it imparts frequency variation to the sound.

Although well received by churches, the signature B-3 Chorus/Vibrato graced jazz and rock recordings ever since. Now you can use this beautiful effect on any instrument.

How Voce Chorus/Vibrato is Used

Simply click the Big Knob to rotate between settings of Vibrato and Chorus. V1 provides the least amount of vibrato, V2 slightly more, and V3 the most. Likewise the amount of Chorus increases from C1 to C3.

Voce Chorus/Vibrato

Option-Click the knob to rotate it in the opposite direction, or click the lettering to select a specific setting.

Tips and Tricks

The classic setting for organ is "C3" but you'll find other settings useful on a variety of instruments. Some of our favorites include:

Electric Pianos

Many electric pianos feature built-in vibrato. But if the sound you're using doesn't provide a realistic vibrato (perhaps you're wrestling with a sampler), track dry and apply the effect later.

Guitar

A certain popular guitar amp has a knob that says "Vibrato" but it's really just Tremolo. Tremolo is amplitude modulation; the sound gets louder and quieter. Vibrato, in contrast, imparts pitch change. A select few highly sought after '50s Magnatone guitar amps feature a true tube vibrato (one even does stereo!) You can approximate this sound by recording guitar direct (or starting with a clean miked sound), applying Voce Chorus/Vibrato, then using SansAmp™ PSA-1.

Voce Spin

(TDM, RTAS, and AudioSuite)

Voce Spin provides the most accurate simulation of the well-loved rotating speaker. 15 classic recording setups feature horn resonance, speaker crossover, varying microphone placement—even the "Memphis" sound with the lower drum's slow motor unplugged!

How Voce Spin Works

Don Leslie invented the rotating speaker in 1937. His design is simple and elegant: an internal 40-watt tube amplifier feeds a speaker crossover, which splits the signal.

All frequencies below 800 Hz go to a 15" bass speaker and all frequencies above 800 Hz go to a compression horn driver.
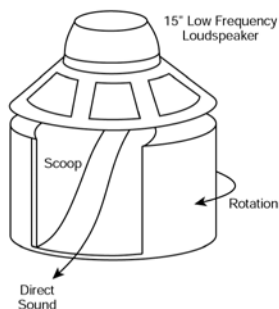
Lower speaker assembly

The large bass speaker is bolted to the cabinet and a foam drum directly below the speaker reflects the bass outward.

For the high frequencies, a treble horn with two bells reflects the sound from the compression horn driver located below.

Only one bell actually produces sound; the other is merely a counterbalance.
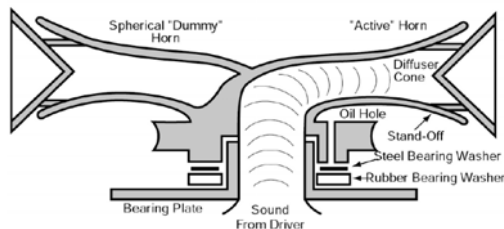
Upper speaker assembly

Then, of course, it spins. Separate belts, pulleys and motors drive the upper treble horn and the lower foam drum. Adding to the effect, the upper horn and lower drum spin in opposite directions. Most rotating speakers feature two sets of motors, allowing both slow ("Chorale") and fast ("Tremolo") rotation speeds.

How Voce Spin is Used

Of course, all that motion creates a rich sound—but then you have to capture it using microphones. Spin provides fifteen classic recording setups to choose from, giving you the sounds you've heard on countless records instantly.

Voce Spin

Just choose a preset and click Chorale, Tremolo, or Off. Alternately, click and drag the flip switch. Short flicks of the wrist land on Off; longer flicks toggle between Chorale and Tremolo.

🔆 *See also "Additional Controls" on page 48.*

You may also Alt-click (Windows) or Option-click (Mac) anywhere to toggle between Chorale and Tremolo speeds.

Spin Presets

122 Model 122 speaker, medium pulleys

122 (small pulley) Small pulleys (fastest rotation)

122 (large pulley) Large pulleys (slowest rotation)

122 (wide stereo) Middle pulleys, wide stereo microphone placement

122 (mono) Middle pulleys, one mic each top and bottom

21H Model 21H speaker

Foam Drum Middle pulleys, microphones close

Memphis Lower drum slow motor unplugged, microphones close

Steppenwolf Lower drum only, loose belts, microphones close

Rover Slow to FastGuitar rotating speaker, maximum speed differential

Rover Slow to MediumGuitar rotating speaker, slower variation

Rover Medium to FastGuitar rotating speaker, faster variation

Phaser Medium rotation rate, microphones very close

Watery Guitar Fast rotation rate, microphones close

Speed Options

Chorale Slow rotation

Tremolo Fast rotation

Off No rotation, but still through the crossover and speakers (wherever the speakers comes to rest relative to the microphones!)

Additional Controls

Though the Voce Spin plug-in window contains only the Chorale/Off/Tremolo control, the following parameters are also available:

- Input Trim
- Speed Switch
- Rotor Balance
- Upper Slow Speed
- Upper Accel Rate
- Upper Decel Rate
- Upper Mic Angle
- Lower Fast Speed
- Lower Slow Speed
- Lower Accel Rate
- Lower Decel Rate
- Lower Mic Angle

Using these controls, you can adjust and automate parameters such as input trim (from –24 dB to +24 dB), set the rotor balance (the mix between the upper and lower speakers), specify acceleration and deceleration times (in seconds) for both the upper and lower speakers, tweak the fast and slow speeds of each speaker, and specify the microphone angle for each stereo pair of microphones.

You can access these additional controls through Pro Tools plug-in automation, and/or from a compatible control surface.

Accessing Controls On-Screen

All Voce Spin parameters can be adjusted on-screen by editing Pro Tools breakpoint automation data.

To access additional Voce Spin parameters on-screen:

1 Click the Auto button in the Voce Spin plug-in window to open its Plug-in Automation window.

2 In the list of parameters at the left, click to select a parameter and click Add (or, simply double-click the desired parameter in the list). Repeat to access and enable additional controls.
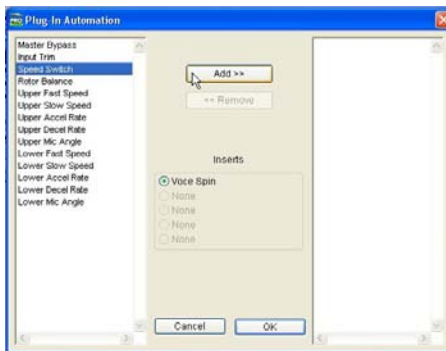
Enabling a parameter in the Plug-in Automation window

3 Click OK to close the Plug-in Automation window.

4 In the Edit window, click the Track Display selector and choose the automation parameter you just enabled from the Voce Spin sub-menu.

5 Edit the breakpoint automation for the enabled control.

Accessing Controls from a Control Surface

When using a control surface, all Voce Spin parameters are available whenever the plug-in is focused. You only need to enable plug-in automation (as described previously) if you want to record your adjustments as breakpoint automation.

To access additional Voce controls from a control surface:

1 Focus the Voce Spin plug-in on your control surface. All available parameters are mapped to encoders, faders, and switches.

2 Adjust the control currently targeting the desired parameter.

3 If necessary, use the previous/next Page controls to access additional controls.

💡 *To automate your adjustments, be sure to enable automation for that parameter as described above. See the Pro Tools Reference Guide for complete track automation instructions.*

Tips and Tricks

The "One Mic Way Back In The Corner Of The Room" Trick

Spin isn't designed to sound like a rotating speaker spinning all by itself in a large room. Spin provides the sound of a miked rotating speaker, the sound the producer and engineer hear in the control room. But don't let that stop you from getting the sound you want!

To achieve the sound of a distant microphone capturing the rotating speaker, run Spin using the wide stereo preset. Now apply a room reverb, remove any pre-delay, and adjust the wet/dry reverb balance until you get the distant sound you're looking for.

Spin into Moogerfooger Lowpass Filter

Try using the amplitude modulation effects of Spin as an LFO driving the Moogerfooger Lowpass Filter!

Distortion and Spin

To simulate overdriving the tube amp powering the rotating speaker, apply distortion before Spin, since, in the real-world signal path, the amp distorts the signal before the speakers throw the sound around. Among tons of other great distortion sounds, the SansAmp PSA-1 plug-in provides distortion presets for both the model 122 and model 147 rotating speakers.

Organ Signal Path

Likewise, when going for classic organ sounds, route through the Voce Chorus/Vibrato before Spin, as that's the signal path in the B-3 organ.

The John Lennon Vocal Thing

In what seems like a particularly dangerous Beatles studio experiment, a Leslie speaker cabinet was dismembered, a microphone was affixed to the rapidly spinning upper rotor, and John Lennon attempted to sing into it. Fortunately the deafening wind noise captured by the microphone put a stop to the proceedings before anyone got maimed. Feel free just to run the vocal through the rotating speaker—that's what they wound up doing.

Reverse Spin

Those reverse-vocal and reverse-guitar tricks are even more fun when you run 'em through Spin. Try reversing the vocal and putting it through Spin, as well as putting the vocal through Spin then reversing the processed vocal.

Generator Leakage

Of all the sounds to pass through a Leslie, no sound has been amplified more often than the sound of B-3 Organ generator leakage. Even with no notes keyed, a small amount of B-3 sound leaks out. We've provided a sample on the Installer disc—run it through Spin and keep a keen eye out for ghosts.

Cosmonaut Voice

Cosmonaut Voice

(RTAS and AudioSuite)

The Cosmonaut Voice plug-in is a radio and shortwave simulator. Use it to add squelch or noise to tracks.

Cosmonaut Voice

How the Cosmonaut Voice is Used

Cosmonaut Voice is, in simple terms, an amplitude-driven noise generator with adjustable sensitivity, selectable noise type (beep or squelch), and an additional RFI/static noise generator.

Cosmonaut Voice provides controls on-screen for Threshold, Noise, and Beep/Squelch.

Threshold Sets the point at which the selected voice (beep or squelch) is triggered. Turning Threshold clockwise raises the threshold and increases sensitivity (resulting in more triggering); turning Threshold counter-clockwise decreases sensitivity.

Noise Raises or lowers the amount of RFI/static noise mixed in with the signal (independent of the Beep/Squelch or Threshold controls). Turning Noise to the right adds a more constant noise "bed" behind the Beep/Squelch effect; turning Noise to the left decreases the ambient noise, resulting in sharper Beep/Squelch cut-in.

Beep/Squelch Sets the voice mode between Beep (NASA-style radio beep) and Squelch (noise burst).

Accessing Additional Controls On-Screen

Cosmonaut Voice also provides a
Beep/Squelch Level control to set the balance of
the generated noise and dry signal.
Beep/Squelch level can be adjusted on-screen by
editing Pro Tools breakpoint automation data.

To access Beep/Squelch level on-screen:

1 Click the Auto button in the Voce Spin plug-
in window to open its Plug-in Automation win-
dow.

2 In the list of parameters at the left, click to se-
lect B/S Level and click Add (or, simply double-
click the desired parameter in the list). Repeat to
access and enable additional controls.
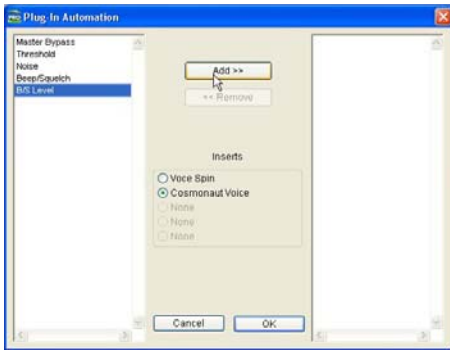
Enabling a parameter in the Plug-in Automation window

3 Click OK to close the Plug-in Automation win-
dow.

4 In the Edit window, click the Track Display se-
lector and choose B/S Level from the Cosmo-
naut Voice sub-menu.

5 Edit the breakpoint automation for the en-
abled control.

Accessing Controls from a Control Surface

When using a control surface, all plug-in param-
eters are available whenever the plug-in is fo-
cused. You only need to enable plug-in automa-
tion (as described previously) if you want to
record your adjustments as breakpoint automa-
tion.

To access the Beep/Squelch level from a control surface:

1 Focus the Cosmonaut Voice plug-in on your
control surface. All available parameters are
mapped to encoders, faders, and switches.

2 Adjust the control currently targeting the de-
sired parameter.

*To automate your adjustments, be sure to
enable automation for that parameter as
described above. See the Pro Tools Reference
Guide for complete track automation in-
structions.*

Fairchild Plug-ins

This chapter describes the following Fairchild plug-ins:

- Fairchild 660 (see "Fairchild 660" on page 53)
- Fairchild 670 (see "Fairchild 670" on page 55)

Fairchild 660

(TDM, RTAS, and AudioSuite)

Re-introducing the undisputed champion in price, weight and performance: the $35,000, one-hundred pound, Fairchild 660.

Bomb Factory's no-compromise replica captures every detail of this studio classic.

How the Fairchild 660 Works

Designed in the early 1950s, the Fairchild 660 is a variable-mu tube limiter. Variable-mu designs use an unusual form of vacuum tube that is capable of changing its gain dynamically.

The result? In addition to featuring a tube audio stage like the LA-2A, the Fairchild actually achieves gain reduction through the use of tubes!.

Tubes, wires, and iron

The heart of the Fairchild limiter—a 6386 triode—is one such variable-mu tube. In fact, four of these tubes are used in parallel! A key part of the Fairchild design, it ensures that the output doesn't get darker as the unit goes further into gain reduction, and also reduces distortion as the tubes are biased further into Class-B operation.

How the Fairchild 660 is Used

The Input Gain control sets the input level to the unit and the compression threshold, just like the Input control on an 1176. Full clockwise is loudest.

Bomb Factory Fairchild 660

The Threshold control adjusts the gain to the sidechain, just like the Peak Reduction control on an LA-2A.

Adjust these two controls together until you get the sound you want. Like many classic compressors, after a little bit of tweaking, you'll know immediately when you get it right.

The Time Constant knob selects the attack and release times for the compressor. One is fastest, and six is slowest. Seven and eight are Bomb Factory custom presets.

Tips and Tricks

5,6,7,8...

The Fairchild manual documents Time Constant settings 5 and 6 as user presets—although you have to go inside with a soldering iron to change them. We used the "factory default" values.

Bonus Settings

Settings 7 and 8 do not exist on real-world units—well, at least most of them. These settings are taken from a real-world Fairchild modification invented by Dave Amels many years before he designed the plug-in version.

What do they do? Settings 7 and 8 offer versions of Time Constant 2 with a gentler release useful for compressing vocals and other program material where you desire more subtlety in the compression action. Give them a try—you've already heard them on hit songs on the radio.

Pump It Up

With a carefully adjusted Input Gain and Threshold, you can use Time Constant 1 to achieve a cool pumping effect on drums. The sound gets darker and fuller, and sits beautifully in a mix.

Fairchild 670

(TDM and RTAS)

Just like the Fairchild 660, Bomb Factory's no-compromise replica captures every detail of the Fairchild 670.

The Fairchild 670 is a dual-channel unit and, as such, is only available on stereo tracks.

Note that the companion Fairchild 660 also supports stereo operation. Bomb Factory modeled both a Fairchild 660 and a Fairchild 670 from scratch using two different hardware units. This gives you a choice of two different-sounding Fairchild units to try on your stereo tracks!

How the Fairchild 670 Works

The Fairchild 670's internal design is similar to the Fairchild 660. However, the Fairchild 670 offers two channels of compression instead of one. Combined with the Lat/Vert mode described later, this gives you even more compression options on stereo tracks.

Fairchild 670

How the Fairchild 670 is Used

The Input Gain control sets the input level to the unit and the compression threshold, just like the Input control on an 1176. Full clockwise is loudest.

The Threshold control adjusts the gain to the sidechain, just like the Peak Reduction control on an LA-2A.

Adjust these two controls together on both channels until you get the sound you want. Like many classic compressors, after a little bit of tweaking, you'll know immediately when you get it right.

The Time Constant knob selects the attack and release times. One is fastest, and six is slowest. Seven and eight are Bomb Factory custom presets. See "Fairchild 670" on page 55 for details on these custom settings.

The AGC knob lets you select Left/Right processing or Lat/Vert processing of the two channels. Left/Right works like a dual-mono compressor with separate controls for the left and right channels. In Lat/Vert mode the top row of controls affects the in-phase (Lat) information and the bottom row of controls affects the out of phase (Vert) information. Although originally designed for vinyl mastering where excess Vert (vertical) information could cause the needle to jump out of the groove, you can use the Lat/Vert mode to achieve some amazing effects – especially on drums.

Tips and Tricks

To exactly match the settings between channels, hold down the Alt key (Windows) or the Shift key (Mac) while dragging the mouse. This is useful when trying to preserve the existing Left/Right balance on stereo material.

chapter 17

Pultec Plug-ins

This chapter describes the following Pultec EQ plug-ins:

- Pultec EQP-1A (see "Pultec EQP-1A" on page 57)
- Pultec EQH-2 (see "Pultec EQH-2" on page 59)
- Pultec MEQ-5 (see "Pultec MEQ-5" on page 60)

Pultec EQP-1A

(TDM, RTAS, and AudioSuite)

The Pultec EQP-1A provides smooth, sweet EQ and an extremely high quality tube audio signal path. Use it on individual tracks, critical vocals, or even across a stereo mix for mastering applications.

How Pultec EQP-1A Works

Built in the early 1960s, The Pultec EQP-1A offers gentle shelving program equalization on bass and highs, and offers a variable bandwidth peak boost control. A custom (and secret) filter network provides all its equalization functionality. Quality transformers interface it to real-world studio equipment. And a clean and well-designed tube amplifier provides a fixed amount of make-up gain.

Pultec EQP-1A

How Pultec EQP-1A is Used

Low Frequency Section

Adjust low frequencies using the Boost and Atten knobs and the Low Frequency switch, located at the left side of the unit. All low-frequency equalization is a gentle shelving type, 6 dB per octave.

High Frequency Boost Section

Boost mid and high frequencies using the Bandwidth and Boost knobs and the High Frequency switch.

High Frequency Attenuate Section

Cut high frequencies using the Atten knob and the Atten Sel switch located at the right side of the plug-in.

Tips and Tricks

Twelve O'Click

Alt-click (Windows) or Option-click (Mac) any knob to reset it to its unity position.

"Q" and A

You may wonder why the Pultec EQP-1A has separate knobs for boost and cut. The short answer is that they connect to different circuitry in the unit.

You can use the "extra" knob to your advantage. Because the filters are not phase perfect, a Boost setting of 3 and an Atten setting of 3 can make a huge difference, even though a frequency plot wouldn't show much difference in tone. You're hearing the phase shift, not the tone shift.

Our ears of very sensitive to phase, and using the two knobs together, you can adjust phase at the low end while also making tonal adjustments.

On the high end, you can set Boost to 10k and Atten to 10k, then adjust Boost and Atten simultaneously. However, because Boost is a peak equalizer and Atten is a shelving equalizer, the results are much different, and you don't get independent control of phase.

"Q" and Boost

In the high frequency boost section, the Bandwidth and Boost controls affect one another. This is different from modern equalizers, where adjusting Q typically doesn't affect the amount of equalization applied.

Use caution, because the Sharp bandwidth setting results in up to 10 dB higher output than Broad bandwidth at maximum Boost, just like on the original. But don't feel like you're getting cheated. Here at Bomb Factory, we consider anything that encourages very careful and infrequent use of peaky boosts to be a Very Good Thing.

Pultec EQH-2

(TDM, RTAS, and AudioSuite)

The Pultec EQH-2 is a program equalizer similar to the Pultec EQP-1A. It is designed to provide smooth equalization across final mixes or individual tracks.

How Pultec EQH-2 Works

The Pultec EQH-2 offers three equalization sections: low frequency boost and attenuation, midrange boost only, and 10k attenuation. Like its EQP-1A sibling, it features high-quality transformers and a tube gain stage. But unlike the EQP-1A, the tube stage in the EQH-2 is a push-pull design. As a result, the EQH-2 offers a beefier tone.

Pultec EQH-2

How Pultec EQH-2 is Used

Low Frequency Section

Adjust low frequencies using the top row of Boost and Atten knobs and the CPS (cycles per second) switch. All low-frequency equalization is a gentle shelving type, 6 dB per octave.

High Frequency Boost Section

Boost mid and high frequencies using the KCS (kilocycles per second) and Boost knobs on the second row.

High Frequency Attenuate Section

Cut high frequencies using the 10k Atten knob located at the right side of the plug-in.

Tips and Tricks

Alt-click (Windows) or Option-click (Mac) any knob to reset it to its unity position.

Pultec MEQ-5

(TDM, RTAS, and AudioSuite)

The Pultec MEQ-5 is the most unique equalizer in the Pultec family. It is particularly useful on individual tracks during mixdown.

How Pultec MEQ-5 Works

The Pultec MEQ-5 offers three equalization sections: low frequency boost, mid frequency boost, and wide-range attenuation. Like all Pultecs, it features quality transformers and a tube gain stage.

Pultec MEQ-5

How Pultec MEQ-5 is Used

Low Frequency Peak

Boost low frequencies (200, 300, 500, 700, 1000 Hz) using the upper left controls.

Mid Frequency Peak

Boost mid-frequencies (1.5k, 2k, 3k, 4k, 5k) using the controls at the upper right.

Wide-Range Dip

Cut frequencies using Dip controls on the bottom row.

Tips and Tricks

Guitars

Have multiple guitars that sound like mush in the mix? The Pultec MEQ-5 is a classic tool for achieving amazing guitar blends. Try boosting one guitar and cutting another to achieve an octave of separation. For example, cut one guitar using 1.5 (1500 Hz) Dip, then boost the other using 3 (3000 Hz) Peak. View the matched pairs of presets (Guitar 1A and 1B, 2A and 2B, etc.) for further examples of this technique.

Twelve O'Click

Alt-click (Windows) or Option-click (Mac) any knob to reset it to its unity position.

chapter 18

Slightly Rude Compressor

Slightly Rude Compressor

(TDM, RTAS, and AudioSuite)

The Slightly Rude Compressor is Bomb Factory's first completely custom compressor design. Used conservatively, it sounds beautiful on vocals, drums, guitars and piano. Pushed hard, it's unique and aggressive. The stereo version is specifically designed to solve the problems that often plague digital mixes.

How the Slightly Rude Compressor Works

The Slightly Rude Compressor is not based on any specific piece of vintage hardware. It is a completely custom design that features all of Bomb Factory's knowledge and expertise in the realm of digital compression.

Slightly Rude Compressor

How the Slightly Rude Compressor is Used

The Input Amount control sets the input level to the unit and the compression threshold, just like the Input control on an 1176. Full clockwise is loudest.

The Make-Up Gain control adds gain after compression. It works just like the Gain control on an LA-2A.

The Release Time adjusts the release time; full clockwise is fastest and provides the most "pump."

The Rudeness knob and Slightly Rude/Super Rude switch affect the sound of the compression action.

Tips and Tricks

For a classic sound, use the "Slightly Rude" setting and keep the Rudeness control below the half-way point. Settings above 50% will increase the aggressiveness of the compressed sound.

To achieve more dynamic effects, switch to the "Super Rude" mode. In this mode, the Rudeness knob controls the amount of overshoot in the compressor. This results in a distinctive processed sound on percussive material, especially on piano and drums.

Try chaining the Slightly Rude Compressor either before or after other compressors. Using the Bomb Factory Fairchild 660 (or 670) or Bomb Factory BF76 before or after the Slightly Rude Compressor will give you an amazing variety of compression options – especially if you experiment with the Super Rude mode.

Also be sure to try the Slightly Rude Compressor on full mixes or stereo submixes! It adds the "glue" that helps hold mixes together, something that's often hard to achieve in the digital domain.

All About Compressors

This appendix provides an overview of dynamics processing for general reference, and for specific examples of how some of the Bomb Factory dynamics plug-ins can be utilized.

What Is Compression?

The difference between the softest and loudest portion of an audio signal determines its dynamic range. Compressors allow you to control the dynamic range of a signal.

Common uses for compression include smoothing out vocals, adding punch to bass tracks, or taming that drummer who gets a little too excited with the kick drum right before each chorus. Used as a de-esser, compression also assists in removing excess sibilance from vocals.

How a Compressor Works

The heart of a compressor lies in its detector and voltage-controlled-amplifier (VCA). The detector monitors the output and calculates the proper gain reduction. It then passes that information to the VCA, which automatically "rides the gain" of the input signal, decreasing the louder peaks.

Detectors use various electronic circuits. Some respond instantly to program peaks, making them useful for "brickwall" limiting in audio broadcast or mastering applications. Other circuits perform simple averaging or more natural sounding RMS calculations that are better suited to a wider range of program material.

Once the detector determines the output level, the VCA adjusts the signal level. VCAs vary in type from cheap operational amplifiers (op amps) to elaborate tube circuitry. Because the VCA is a gain element in the signal path, it greatly impacts the perceived quality of the final sound. In a true compressor, the VCA only reduces the input signal and, as a result, you need to add gain afterward to bring the average signal back to the original perceived level.

The 1176 compressor allows you to select a compression ratio. Compression ratio determines how the input signal relates to the output signal. If for each 4 dB increase in input level above a threshold the output level only increases by 1 dB, the compression ratio is 4:1. Small ratios (like 2:1 or 4:1) add subtle polish; higher compression ratios (like 20:1) produce a distinctly "compressed" sound.

Most modern compressors also feature a sidechain input. A sidechain allows you to feed a separate audio signal directly into the detector, while still applying compression to the original input signal. Although neither the 1176 nor the

LA-2A originally had sidechains, many were put under drill presses to accommodate the wishes of audio engineers. Both the Bomb Factory BF76 and Bomb Factory BF-3A offer sidechain capability.

How Compression is Used

Restrict Dynamic Range

The simplest use of a compressor involves taming the louder peaks of an input signal. The 1176 and LA-2A are very easy to use—set the Output Level ("Gain" on the LA-2A) to a comfortable setting, then adjust Input Level ("Peak Reduction" on the LA-2A) until you clearly hear the compression action.

If the sound is too quiet, muffled, or dead sounding, you're probably using too much compression. But if you notice motion around the louder peaks in the program material (either a "squashed" sound or a "pumping" effect), you're getting close.

Because the compression action decreases the overall signal level, adjust the Input/Gain control to smooth out the level differences. Then readjust Peak Reduction as necessary for the desired sound.

To get a better understanding of what's going on, look at the meter. The typical setting is "Gain Reduction" ("GR" on the 1176). In this mode, the compressor's VU meter moves backward from 0 VU to show the amount of gain reduction being applied to the signal. To get a feel for each compressor, adjust so the needle bounces backward about 10 dB on each peak, and listen for the effect.

Breathe Life Into Lame Tracks

What is the correct way to use compression? It depends on the desired effect and the problem at hand. Smoothing the overall dynamics of a performance often increases the strength and presence of a track in the final mix. On the other hand, misuse of compression can suck all the life out of a performance, making highly marketable recordings for radio airplay in certain decades. Which is to say that there's no wrong way to use compression, just wrong ways right now.

Let's suppose it's 3:00 AM and your mix just isn't happening. The CD burner blinks ominously, waiting to be fed the shiny master disc due at 9:00 AM. You solo each track one more time. And there it is: the bass. It sucks. Undeniably, it sucks. Never has a bass sucked like this bass sucks.

Depressed, you analyze the situation. In addition to being incredibly lame, the bass has too much high end, the attack doesn't cut through, and the tail end of long notes disappears into the mix. A perfect candidate for the LA-2A compressor! The LA-2A is slow enough to respond that the attack will beef up and the tails will smooth out. Plus, if you squash it hard and crank up the gain, the LA-2A fattens up the sound. Problem solved. (But don't forget to fire the bass player.)

De-Essing

Using the compressor's sidechain, you can remove unwanted sibilance from vocals. Now it's 8:00 AM and, after a brief nap, you notice that the backing vocals are ripping your ears off. Since you ran out of tracks about six hours into the process, you've only got the stereo submix to work from, so there's no hope for manually fading the "S" sounds on the individual tracks. So fire up the BF76—it's faster than the BF-2A and more suited to this task. Instead of using EQ to cut the frequencies on the input signal, use EQ to boost the problem frequencies feeding the sidechain. The BF76 compressor responds by compressing more of the problem frequencies, removing unwanted sibilance from the output.

Un-Pumping Drum Tracks

Using a similar technique, you can remove unwanted pumping from drum tracks. Perhaps you're doing a submix of drum tracks and you want to compress the submix. One thing to listen for is any unwanted "pumping"—typically low frequency energy sucking down the high end.

Suppose the drummer is using mallets on a ride cymbal while playing a kick drum. Because of the enormous bass energy, a side effect of the compression is the cymbal getting quieter during each kick. Depending on the compression settings required for the remainder of the program material, this may or may not produce the desired result.

Similar to the De-Essing process, if you EQ the sidechain and reduce low frequency energy (perhaps shelving everything below 300 Hz by 12 dB), there's less low frequency feeding the detector. You gain flexibility in the amount of compression you can use, plus the option of using faster attack times without introducing distortion.

Pre-Compression

There's a lot of marketing-speak about "pumping and breathing" out there. Since they're modeled after the original devices, Bomb Factory dynamics plug-ins "pump and breathe" just like the real ones do. Think of it as the old fashioned pumping and breathing that started this whole mess.

If you want more of an exaggerated effect, try timeshifting the sidechain signal. Simply duplicate the track you're compressing and shift it back in time. (Or get fancy with a delay and signal routing—whatever works for you.) Then take the earlier signal and use it to feed the sidechain input.

Depending on the amount of pre-delay, you can achieve instantaneous attack times (a very fresh sound on the LA-2A) or even unusual "pre-compression" sounds where you hear the compression before you hear the louder program material that caused the compressor response.

index

Numerics

660 (Fairchild Limiter) 53

A

Activation Code 6
adjusting plug-in parameters
    computer keyboard 10
    keyboard shortcuts 9
    scroll wheel 10
    toggling switches 10
age (simulating effect of) 19
aggressive tone 25
allpass 39
Amels, Dave 54
amp simulation 43
analog delay (see also delay) 35
analog filters 39
AudioSuite 11
authorizing plug-ins 6, 7
auto wah 42

B

beefier 59
BF Essential Clip Remover 11
BF Essential Correlation Meter 12
BF Essential Meter Bridge 12
BF Essential Tuner 13
BF-2A 27
    side-chain filter 28
BF-3A 25
BF76 15, 31
Bomb Factory BF76 15, 31
boxed version 2
brickwall 63
brighten 23
broadcast limiter 26

broadcast noise 51
Bucket Brigade Analog Delay Chips 36
buzz 44, 51

C

cabinet simulation 43
Carrier Oscillator 38
clarify 23
Class-B 53
clip remover 11
compound release circuit 23
compression 63
compressor
    BF-2A 27
    BF-3A 25
    BF76 15
    JOEMEEK SC2 22
    Slightly Rude 61
computer keyboard
    adjusting plug-in parameters 10
Cosmonaut Voice 51
crackle (adding) 19
crunch 44

N

noise
    adding radio noise 51
    meter 13

O

oscillator 37
overshoot 23, 62

P

package contents 2
Peak 12
phase
    for EQ 58
    metering 12
phaser (phase shifter) 39
plug-in controls
    adjusting 9
plug-ins
    adjusting parameters 9
    authorizing 6
    installing 5
    registration 2
pre-compression 65
program equalization 57
Pultec
    EQH-2 59
    EQP-1A 57
    MEQ-5 60
pumping 54, 65
punch 44

R

radio effects 51
ratio 63
registration 2
removing plug-ins
    Windows 7
resonance filter 41
resonant filters 39
ring modulator 37
RMS 12
Robinson-Dadson 13
Rotor Balance 48

S

SansAmp PSA-1 43
scroll wheel
    adjusting plug-in parameters 10
separation 60
shortwave simulation 51
side-chain 16, 63
    filter in BF-2A 28
sinusoidal 37
Slightly Rude Compressor 61
smooth EQ 57
sound generator 37
squawk and squelch 51
squishy 15
sum and difference frequencies 38
switches
    adjusting plug-in parameters 10

T

Tel-Ray Variable Delay 33
tighten 23
tin 21
tremolo 37, 39
triode 53
tube sounds 43
tubes 53
tuna can 33
tuning (tuner) 13

U

unity 44

V

variable-mu 53
VCA 63
vibrato 37
vinyl mastering 55
VU meters 12

W

wah 42, 44
website 3
whistling 42
whooshing 39

digidesign
A division of *Avid*

www.digidesign.com