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Chapter 1 - Introduction

The Setups Library's purpose

The C1 at heart is simply a compressor and expander/gate with some added filters, but its unique design, evolved through many years of experience and refinement, makes it a versatile tool for any number of specialized and useful audio processing tasks. Moreover, its performance at most of these tasks is state-of-the-art, either comparable to or superior to dedicated professional products available elsewhere. Many of the tasks it can do are unique to the C1.

The C1 Setups Library allows the C1 to be configured to become any of a number of dedicated processors. The Library consists of a variety of setup files, each of which when loaded turn the C1 into a specialized problem solver or creative tool.

What can you do with the C1?

Among the tasks the C1 can perform using the setups library are the following, with relevant setups and their chapters listed:

• Conventional high-level compression

C1 Classic Compressor/Gate
C1 Classic Compressor/Expander
C1 HLcompress + DeHiss
C1 HLcompress+bass/treb enhance
C1 HLcompress+treble enhance

• Conventional Low level expansion

Chapter 3:	C1 Classic Compressor/Expander
Chapter 4:	C1 EQ + LLexpander
-	C1 Multimedia Speech 2

• Versatile conventional gating

Chapter 3:	C1 Classic Compressor + Gate
-	C1 Classic DeEsser + Gate
Chapter 4:	C1 Multimedia Speech 1

• De-esser - reducing excessive speech or vocal sibilance

Chapter 3:	C1 Classic DeEsser + Gate		
Chapter 7	- all setups		

- Noise reducers reducing background noises
 - Chapter 5:C1 HLcompress + DeHissChapter 6- all setupsChapter 7:C1 De-Ess + De-Hiss 1C1 De-Ess + EQ + De-Hiss 2C1 De-Hiss + EQ + De-Ess 3C1 De-Hiss + EQ + De-Ess 4C1 De-Hiss + EQ + De-Ess 4
- De-reverberation reducing the audible effect of room reverberation, especially on speech
 - Chapter 5: C1 Compressor + De-Reverb
- Hiss removal

See Noise Reducers above

- Rumble removal Chapter 6: C1 Rumble Reducer C1 High-pass filter
- Mid-level compressor making sounds louder without compressing the dynamics of loud sounds

Chapter 4:	C1 Multimedia Speech 1	
-	C1 Multimedia Speech 2	
Chapter 5:	C1 Compressor + De-Reverb	
Chapter 8:	C1 MLcompress+bass/treb enhance	
-	C1 MLcompress+treble enhance	

• Low level detail enhancer - bringing up the level of quiet sounds to make them more clearly audible

Chapter 4:	C1 EQ + LLcompressor
Chapter 7:	C1 De-Ess + LLcompress
Chapter 8:	all setups

 Spectral enhancers - giving sounds more depth, brightness or impact without the unpleasant side effects of conventional equalization or the harshness and artificiality of some commercial enhancers.

Chapter 8: all setups

• Dynamic equalization - permitting a sound to be equalized in different ways at different sound levels, giving effects impossible with a simple equalizer. Chapter 8: all setups • Speech/Vocal intelligibility enhancer - makes speech and vocals more clearly audible under difficult listening conditions without sounding artificial under ideal conditions

Chapter 8:

C1 Speech Enhancer

• Louder multimedia files - allows optimum sound quality to be obtained even from 8 bit 22 kHz multimedia files without "squeezing the life out of" the original sound file.

Chapter 4:

C1 Multimedia Speech 1 C1 Multimedia Speech 2

- Ducking of one signal keyed by another Chapter 9: C1 Ducking 1 C1 Ducking 2
- Gating of one signal keyed by another Chapter 9: C1 Keyed Gate
- Expansion of one signal keyed by another ideal for creating convincing rhythmic tracks out of any continuous sound effect. Chapter 9: C1 Keyed Expander
- Duck-EQ keyed ducking within a frequency band of one signal keyed by another, to allow space to be created in a mix when several sounds conflict in the same frequency band.

Chapter 9:

C1	Ducked EQ 1	
C1	Ducked EQ 2	

 Creative keyed equalization of one signal by another, allowing the dynamics of a key signal to alter the sound of a second signal Chapter 9: C1 Keyed EQ Expander

Combinations of effects

Also, the C1 allows one-pass real-time preview and processing of many combinations of two or even three effects at the same time, for example allowing you to compress a sound file and at the same time to remove side-effects such as increased audibility of noise or room reverberation. These combined effects include:

- Compressor + Low-level expander
 - Chapter 3: C1 Classic Compressor + Expander Chapter 4: C1 Multimedia Speech 2
- Compressor + gate Chapter 3: C1 Classic Compressor + Gate Chapter 4: C1 Multimedia Speech 1
- Compressor + equalizer Chapter 4: C1 EQ + LLcompressor C1 Compressor + EQ
- Equalizer + Gate Chapter 4: C1 EQ + Gate
- compressor + noise reducers Chapter 5: C1 HLcompress + DeHiss
- expander + de-hissing Chapter 6: C1 Noise Reducer
- compressor + de-hissing Chapter 5: C1 HLcompress + DeHiss
- compressor + de-esser Chapter 7: C1 Compressor + DeEsser 1
- compressor + enhancer Chapter 8: C1 HLcomp
 - C1 HLcompress+bass/treb enhance
 - C1 HLcompress+treble enhance
 - C1 MLcompress+bass/treb enhance
 - C1 MLcompress+treble enhance
- De-Esser + gate Chapter 3: C1 DeEsser+ Gate
- Enhancer + gate Chapter 8: C1 Bass/treb enhance+Gate C1 Treble enhance + Gate
- compressor + de-reverberation Chapter 5: C1 Compressor + De-Reverb

In most cases, the compression may either be of conventional high-level type, including limiting, or mid level compression leaving the dynamics of the loudest parts of sounds unaffected, but bringing up quieter details.

On using setups

All the setups in this library should be treated as starting points. You will always have to adjust the setups for use with each individual sound file, to match its loudness, spectral content, dynamics, and specific problems it suffers from, as well as the specific results you desire.

For your convenience this manual describes for each setup which controls are most useful and how they should be adjusted to achieve the desired effect. The actual adjustment uses your skill as a listener to decide the most satisfactory effect, and suggestions and hints are for guidance, not limitations on your creativity or skill.

The guidance for each setup describes not just its purely technical use but a little about what kind of effects and side-effects can be expected so that the user can listen for these and make a judgement as to the best trade-off in any particular case.

In many setups you may in particular have to adjust the output gain to prevent overload clipping on particular signals, due to transient overshoots etc. Instructions on how to prevent overload clipping are given in section 2.5 (Output level and clipping) of this manual.

In all setups, you have the option of using or not using lookahead. This will not affect the basic functionality of any setup but may affect fine details of sounds. For further details on the effects of lookahead see the C1 User's Guide section on Lookahead and section 2.6 (Lookahead) of this manual.

This manual is divided into a number of chapters, several of which are devoted to setups performing a particular kind of task.

These specialized chapters may be considered almost as product manuals in their own right, devoted to describing the use of a specialized "product" obtained when the C1 is loaded with a particular kind of setup.

There is however, considerable overlap between chapters, because many of the setups in effect involve two processing effects working together, so that some degree of cross-reference between chapters is necessary when the "other" effect belongs to another chapter.

In using this setups Library manual, it is assumed that you have the basic skills in using the C1 found in the WaveSystem Plug-in General Controls chapter of the C1 User's Guide. If not, please review this chapter!

Chapter 2 - Basic Use of Setups

Quick tour of the C1

This section is not a substitute for the C1 User's Guide, but simply an overview that may be found useful in using the setups library, since extensive reference is made to the module structure of the C1 in setups descriptions.



The C1 user interface is divided into the main areas shown in the above figure.

These areas are:

Two dynamics processing modules, and their controls and metering:

- The Comp/Exp ("Compressor/Expander") module This acts as a compressor or an expander (which may operate at low, middle or high input levels). It includes two bar meters to monitor respectively gain reduction (in red) or increase (in yellow), and input dynamics control level (in blue). The various buttons and Value Windows control the operating parameters, as described by the captions, and in detail in the main C1 User's Guide.
- The Gate/Exp ("Gate/Expander") module This acts as a Gate, or a low-level (downward) expander or low-level compressor. It includes two bar meters to monitor respectively gain reduction (in red) or increase (in yellow), and input dynamics control level (in blue). The various buttons and Value Windows control the operating parameters, as described by the captions, and in detail in the main C1 User's Guide.
- A filter module, including graphical display of filter responses
 This filter module may be used to equalize just the control sidechains of the dynamics processor, or to split
 the audio into two bands only one of which is dynamically processed, or any combinations of the two.
 Filter type, frequency and Q buttons and Values Windows are provided for filter adjustment. A graphical
 display shows the frequency responses of the "active" filter band in red, and the complementary "passive"
 band of the bandsplit in blue. The filters may also be adjusted by clicking and dragging the cross-marker on
 the graph.
- An input/output (I/O) graphical display of dynamic signal levels This displays the way the output level of the two dynamics modules (yellow for the Comp/Exp module, light blue for the Gate/Exp module) varies with input level, and shows the moment-by-moment variations of level on the graph itself. The triangular grab markers below the graph display, and can be used to adjust, the threshold settings of the two dynamics modules by clicking and dragging.
- An input and output signal level control and monitoring area This includes an output level slider, input level controls for left and right channels (which are normally left at 0 dB unless alterations in balance are required), peak-reading output level meters for the 2 stereo channels, and buttons to switch between normal stereo operation and keyed operating modes, and a monitor button to allow auditioning either of the processed audio or of the sidechain or passive bandsplit signal band signals for setting up purposes.
- A number of controls for saving, loading, copying and altering setups
- · Controls for previewing and processing sound files
- A title bar indicating name of loaded setup
- An IDR button for optimizing digital resolution at the 16 or 20 bit wordlength.

Detailed instructions on the use of controls are given in the main C1 User's Guide.

Controls which are recommended for use with setups below are indicated within the text in bold typeface.

Loading setups

Place the setup files you wish to use in one (or more) folder of your choice. We shall call this folder the C1 setups folder, but you may choose any other names, and any location on your hard (or other) disc(s) you wish.

Please be sure to keep safety copies of all setup files in the setups Library in another location (e.g. on a floppy disc) so that you have the original setup files in case you accidentally alter the ones in the C1 setups folder(s).

To load the setup you wish to use, click on the Load button, and select the setup file you wish to use, and either click on the Open button or double click on the setup file name. The setup name will appear at the top of the C1 window, and all buttons, Value Windows and graphical displays will be set up to the pre-set settings in that setup.

You may load two setups in the C1 at a time. The SetupA/SetupB button may be clicked to change from one of these setups to the other. If you load a setup in the setup B position, this will not affect the setup loaded into the setup A position, or vice-versa.

This is useful for making quick switched A/B comparisons between two different setups, for example to check whether one sounds better than another.

The name of the last setup loaded or copied into or saved from Setup A or Setup B will be displayed in the title bar at the top of the C1 window.

Making use of Setups

Once you have loaded a setup, you may alter the value or setting of any control on the C1 in the manner described in the main C1 User's Guide.

In most setups, certain controls are important in adjusting the effects and other controls should not normally be used. In the instructions for each setup, the controls that may be adjusted are indicated in bold type.

The simplest way to develop new customized setups for your own use is to modify the values of controls from a single setup in the Setups Library. However, there is another way of compiling new setups from existing ones, i.e. editing by copying and pasting controls settings from one setup to another. Editing allows you to take "bits and pieces" from different setups and incorporate then all into a single new setup. This, for example, you can take some or all of the Comp/Exp module button and Value Window settings from a first setup, some or all of the Gate/Exp module button and Value Window settings from a second setup, and some or all of the filter button and Value Window settings from a second setup, and some or all of the filter button and Value Window settings from a second setup.

Editing is most useful when the Comp/Exp and Gate/Exp modules are set up to do completely independent tasks. Then you can take one module from one setup and the other from another setup to compile a new combination setup. However, be warned that if both use the filter module (i.e. if neither are in Wideband EQ mode), this will only work if both use the same filter module setup. If either of the modules is in Wideband

mode, this will not be a problem.

You can do editing by compiling the setup you want in setup A and loading the setups you wish to edit from, one at a time, into setup B.

Hint: To save time, load the first setup you are compiling from straight into setup A. Then you do not have to paste its control settings from setup B to setup A.

Saving and Editing setups

After a setup has been altered, according to the instructions given and your judgement, you may wish to save it for future use on similar sound files. This is done by clicking the Save button. You will enter a dialog box which allows you select a folder in which to locate the setup, or even to create a new folder to save it in, according to the usual Apple conventions. You should type the name you wish to give the setup in the "Save file as:" window, and then click the Save button in the dialog box. After you have done this, the name of the setup in the title bar at the top of the C1 window will change to the new setup name.

Warning: Before saving a setup, ensure that the monitor button is in Audio mode, unless you specifically want the setup to load later in another monitor mode.

The simplest way to develop new customized setups for your own use is to modify the values of controls from a single setup in the Setups Library, or from previously prepared setups you may have. However, there is another way of compiling new setups from existing ones, i.e. editing. Editing allows you to take "bits and pieces" from different setups and incorporate then all into a single new setup. This, for example, you can take some or all of the Comp/Exp module button and Value Window settings from a first setup, some or all of the Gate/Exp module button and Value Window settings from a second setup, and some or all of the filter button and Value Window settings from a third setup.

Editing is most useful when the Comp/Exp and Gate/Exp modules are set up to do completely independent tasks. Then you can take one module from one setup and the other from another setup to compile a new combination setup. However, be warned that if both use the filter module (i.e. if neither are in Wideband EQ mode), this will only work if both use the same filter module setup. If either of the modules is in Wideband mode, this will not be a problem.

You can do editing by compiling the setup you want in setup A and loading the setups you wish to edit from, one at a time, into setup B. To paste the settings of a number of control buttons, value widows and/or grab markers from setup B to setup A, proceed as in the following example.

First, in setup B, select the desired buttons, Value windows and/or grab markers by click and holding down the mouse button at a point outside the desired controls (not in a button or Value Window!), and dragging the mouse to form a rectangle intersecting or including just the desired controls as shown in the following illustration for all the Comp/Exp module controls:



Letting go of the mouse button, you will have selected all controls intersected by the rectangle. Then press the c key on the keyboard (NOT control C !) to copy the control settings. Then switch to setup A by clicking the Setup A button, select the same controls as before by click and drag, and then press the v key on the keyboard (NOT control V!) to paste the control settings.

This completes the pasting operation.

You may switch back to setup B by clicking the setup B button, load a new setup from the setups Library into setup B, and paste control setting from another part of this setup into setup A by repeating the above procedure. And so on until you have compiled your new setup in setup A. Then you can save it if desired, or just use it for processing.

The same method can be used to compile a setup into setup B from setups loaded into setup A.

Hint: To save time, load the first setup you are compiling from straight into setup A. Then you do not have to paste its control settings from setup B to setup A.

Output level and clipping

You may sometimes find on particular sounds that use of setups Library setups will cause clipping or overload distortion. Such clipping can be caused by excessive gains or from transient overshoots.

The internal processing in the C1 is designed so that clipping distortion cannot occur in the internal signal processing algorithm, so that the only risk of clipping comes from excessive output gain. Thus all you need do to prevent clipping is to reduce output gain appropriately.



Output meter showing ten clippings and with mouse positioned to reset the overload indication by clicking.

The windows above the output level meters show the number of overload clippings occurring during preview or during processing a file in red. They may be reset before a preview or processing by clicking on them. If, after processing, clipping is shown, you may wish to undo the processing and to re-do it with a lower output gain to prevent clipping, although one or two or very few clippings only may prove not to be audible.

The readings at the bottom of the output level meters shows the maximum peak level achieved during preview or processing. It may be reset before preview or processing by clicking on the meter itself. These readings are useful to allow increase in output level if the sound is undermodulated. If for example, the peak level shown in previewing or processing a file is -4.7 dBFS (relative to digital full scale) in the highest of the two channels, then you may increase the output level by say 4.6 dB without the re-processing of the file (after undoing previous processing) causing overload.

However, be aware that the peak level occurring in processing a whole sound file may not be indicated during preview of just a limited segment of that file. You can check peak levels either by

(i) processing the file, and then undoing the processing after taking meter readings, or

(ii) clicking the playback button so that it is illuminated, and then playing the entire sound file under Sound

Designer II[™], or at least all parts of the sound file which are loud enough to be expected to cause clipping problems.

Hint: Especially when using compression or limiting, you may find that transient peaks are reduced in level, often by 2 or 3 dB, if you process using Lookahead in Yes mode, rather than process without Lookahead. However, this can also subtly affect the sound, so use your judgement here as to whether this is desirable in your case.

Lookahead

The use of the lookahead button will not seriously affect any of the setups in the setups library and may be set to "Yes" or "No" as desired by the user.

To some users, the function of the lookahead button may seem mysterious, as it at first sight seems to do little. Internally, it delays the audio signals just enough to match inevitable delays required to derive the gain control signals used to alter signal gains in compression, expansion and gating. Most analog dynamic processors do not have lookahead. The C1 provides the option of simulating analog processing by switching lookahead out (i.e. the lookahead button displays No). With lookahead in (i.e. the lookahead button displays Yes), generally, the shape of compressed transients is better, with less overshoot (see the illustrations below), and there is less premature gating of initial transients such as sibilants or drumstick sounds.

4 kHz test signal with sudden 20 dB step

Effect of compression on 4 kHz test signal without and with lookahead for limiting at threshold -30 dBFS with 5 msec attack time. Note the reduced overshoot of the transient. The effect is less extreme on most real world audio waveforms!

Thus in most situations, the processing is better behaved with lookahead, and setups in the setups library are loaded with lookahead.

However, the actual sound without lookahead is different - for example, transients often sound brighter if compressed without lookahead, so this may be preferred on the basis of sound. The initial gating of transients like short sibilant sounds may also sometimes be wanted to modify the sound - and again no lookahead would then be preferred.

Also, because lookahead starts modifying the sound slightly before the start of a transient, it can sometimes cause a subtle but noticeable "pre-echo" effect sounding like a slightly disturbing room echo, especially on high-quality speech. If this effect is audible, you may again prefer not to use lookahead.

As seen in the above waveform illustrations taken from Sound Designer II[™], the time delays used to implement lookahead have no effect on the timing of processed files, which remain exactly synchronized to unprocessed files.

Classic processors

This chapter describes a few setups that are what may be termed "classic" dynamic processors, the kind of basic workhorses used for most dynamic processing in the audio industry: the Compressor/Gate, Compressor/ Expander and DeEsser/Gate.

Classic Compressor and Gate

This is a classic (high-level) soft-knee compressor/limiter with a classic gate.

Setup name: C1 Classic Compressor/Gate

The Comp/Exp module is here set up as a wideband high-level compressor, and the Threshold, Ratio, Attack and Release controls are conventional in use, provided that the ratio is kept within the range 1 to 50. Above a ratio of 20, the compressor acts as a soft-knee limiter. The Makeup gain allows extra gaiun to compensate for any loss of level due to the compressors reduction of high-level gain.

The PDR (program-dependent release) allow the compressor to respond more rapidly to transients of limited duration, giving a faster release for short transients. The setting here is the duration in msec of transients for which the release time is more rapid. For morte sustained sounds of longer duration, the release time is that set by the Release control. PDR minimizes prolonged gain reductions caused by short transients.

For A/B comparisons of the processed with the original signal, the gain of the bypassed signal may be adjusted by using the bypass gain value window (below the bypass button) so that there is little audible gain change when switching between bypass in and out.

The Gate/Exp module is set to a classic fully-functioned gate. As loaded, the GateOpen and GateClose levels are pre-set 4 dB apart to minimize gate "chatter". The hold control ensures a minimum gate on time which again helps prevent chatter. Attack and Release may be adjusted in the usual way. If it is desired to retain some of the background atmosphere during quiet passages, the floor control may be used to adjust the remaining signal level when the gate is "off".

The Compressor may be switched in or out by clicking on the Comp/Exp bypass button (over the Comp/Exp bar meters).

The Gate may be switched in or out by clicking on the Gate/Exp bypass button (over the Gate/Exp bar meters).

Classic Compressor and Expander

This is a classic (high-level) soft-knee compressor/limiter with a classic soft-knee 2:1 downward expander.

Setup name: C1 Classic Compressor/Expander

The compressor is identical to that in the previous setup, and used in the same way.

The Gate/Exp module is now in low-level (downward) expander mode. GateOpen controls the expander threshold. Attack and Release may be adjusted in the usual way. If it is desired to retain some of the background atmosphere during quiet passages, the floor control may be used to adjust the remaining signal level. The expander is a soft-knee 2:1 device, but the effective expansion ratio below threshold may be increased by using the "negative" polarity settings of the floor control (indicated by "N" after the dB indication in the floor window), with increased effective ratio as the floor setting moves between -100 N and -10 N.

The Compressor may be switched in or out by clicking on the Comp/Exp bypass button (over the Comp/Exp bar meters).

The expander may be switched in or out by clicking on the Gate/Exp bypass button (over the Gate/Exp bar meters).

Classic De-Esser and Gate

This is a classic de-esser with a classic gate.

Setup name: C1 Classic DeEsser + Gate

The Comp/Exp module here is used to implement a de-esser to reduce the level of high-level sibilants. This is implemented as a limiter responding to the level of an equalised sidechain signal in the "Ess" frequency band. The attack and release times are intentionally short to minimise the effect on the signal once an "Ess" sound is over. The degree of de-essing ixs adjusted by moving the threshold up or down. The only other controls used for de-essing may sometimes be the frequency to tune the precise band of "Ess" frequencies.

The gate is identical to that in the C1 Classic Compressor/Gate setup above.

The De-Esser may be switched in or out by clicking on the Comp/Exp bypass button (over the Comp/Exp bar meters).

The Gate may be switched in or out by clicking on the Gate/Exp bypass button (over the Gate/Exp bar meters).

About Compression

Compression is the process of reducing the dynamic range of a sound, by altering the gain. So, for a given number of dB change of level at the input, a smaller number of dB change of level takes place at the output. The ratio

Input dB change Output dB change

is termed the compression ratio, so that if every 2 dB input change causes a 1 dB output change, the compression ratio is 2.

Conventional "hard knee" compressors normally have a constant compression ratio when the input signal rises above a user-preset "threshold level, but the C1 has a more sophisticated compression characteristic matched better to the way the ears hear sound, rather than to an abstract mathematical law. This is not only a "soft knee" law - designed to be neither too hard nor too soft, but involves other features allowing uses in different kinds of compression modes.

The C1 allows three distinct kinds of compression of sounds -

- conventional high-level compression, for which the highest sound levels above a threshold are compressed and reduced in level.
- mid-level compression, in which sounds in the middle of the dynamic range are compressed, but where both very low level and very high level sounds are not compressed.
- low-level compression, for which the very quietest parts of a sound are raised in level to make them louder or more audible, but the louder parts of a sound are unaffected.





Typical high-level compressor setup.



Typical mid-level compressor setup.



Typical low-level compressor setup.

-These three kinds of compression all have important uses, and this flexibility of the C1 makes it a much more effective tool for obtaining precisely the kind of compression needed in any given application. Using conventional high level compression in an attempt to solve all problems often results in an over-compressed sound which has "all the life squeezed out of it", whereas choosing the appropriate form of compression can result in a much more pleasant and natural sound.

Setups for all kinds of compression in the C1 are available. In each case, the other dynamic module can be used for other tasks, often to solve problems that arise when compression is applied. Compressed sounds often have excessively audible low-level noise, or other low-level sounds such as room reverberation or echoes can be increased to an annoying level. The other processing module can often be used to counteract these problems.

Setups in the Library using the three types of compression include:

High-level compression: C1 Classic Compressor/Gate C1 Classic Compressor/Expander C1 Classic DeEsser + Gate C1 Compressor + EQ C1 HLcompress + DeHiss C1 HLcompress+bass/treb enhance C1 HLcompress+treble enhance Mid-Level compression: C1 Multimedia Speech 1 C1 Multimedia Speech 2 C1 Speech Compress/Expand 1 C1 Compressor + De-Reverb C1 Compressor + DeEsser 1 C1 MLcompress+bass/treb enhance C1 MLcompress+treble enhance

Low-Level compression: C1 EQ + LLcompressor C1 De-Ess + LLcompress

In general, wherever a wideband high-level compressor is used in the Comp/Exp module, it may be replaced by a mid-level compressor, and vice-versa, by pasting Comp/Exp control settings from one setup to another.

Which type of compression?

The choice of type of compression depends on what you wish to achieve, and also the nature of the sound file.

If the aim is simply to get the loudest sound possible, then high-level compression or limiting should be used. But be aware that this can also cause the compressed sound to lack a sense of dynamics and seem oppressive. Additionally, especially over cheap reproducing systems with small loudspeakers (e.g. portable radios, portable headphone stereos and small computer loudspeakers), the high average levels can drive the amplifiers or loudspeakers into distortion, giving a fatiguing and unclear sound for the listener. If however, the aim is not loudness for its own sake, but to ensure that the quietest passages are clearly audible

even under difficult or noisy listening conditions (e.g. in an automobile) or via systems with a high floor noise or distortion level (e.g. 8 bit multimedia files), then what you may need is low-level or mid-level compression. This raises the quieter passages above the low level at which they are not adequately audible. This will make those passages clearly audible and less prone to low-level noise or distortion without losing the sense of high-level dynamics, and without pushing all louder sounds to near peak level.

Mid-level compression combines these virtues with making the wanted sound seem louder, by raising the level of the middle sound levels at which signals such as speech spend most of the time.

Low-level compression is preferred where you do not necessarily wish to make the louder sounds in a sound file louder still, but where you need to make the quietest passages or details of sound more audible. Classical music, or drama or documentary where background ambiences need to be made more audible, particularly can benefit from low-level compression.

Chapter 4 - Simple Setups

Multimedia Speech setups

These setups are wideband compressors + gates or downward expanders suitable for use when preparing multimedia speech files, especially those that will later be reduced to 8 bit resolution. This setup may also be used with other multimedia sounds and with speech in other applications where loudness must be combined with quietness between words.

With 8 bit multimedia files, the 8 bit quantization introduces several quality problems. If the sound is above the quantization "floor" level, the quantization sounds like a steady hiss, and it is desirable to minimize its subjective level by maximizing speech loudness. However, if this is done, any background noise is brought up in level, and is liable to give cause the quantization noise to sound very unpleasant between words, with a distinctly "grainy" and intermittent quality. The solution here is to combine mid-level compression to bring up speech levels to be above the quantization noise level during syllables, with gating or effective downward expansion of low-level input signals to prevent them causing the granular sound of bad quantization noise. To minimize loss of quality, this processing should be done on 16 bit files on the C1 possibly at a 44.1 or 48 kHz sampling rate, later using a specialist tool such as the Waves L1 Ultramaximizer to perform the final stage of conversion after sampling rate conversion to maximize file levels and convert to 8 bits.

Soft-knee mid level compression of the kind unique to the C1 is ideal for multimedia speech, since it compresses the middle speech levels at which speech spends most of its time, without "squashing" the dynamics of speech peaks or of low-level sounds. In this way, a more natural loud sound is obtained than by using conventional highlevel compression. The problem that quiet sounds are brought up a great deal in level is solved by the gating or downward expansion.

Undo A: Multimedia Speech		A->B Load Save ? WWAVES	
Comp/Exp Byp 24 0 18 10 20 6 30 0 6 30 40 40 50 40 50 40 12 6 6 40 50 40 50 40 50 40 50 40 10 40 1	Peak Ref Gate Makeup Floor 0.0 -co Threshold GateOpen -45.0 GateClose 2.41:1 Attack Attack Attack 1.50 2.00 Release So 50 25 PDR Hold 10 6.04 EQ mode Wideband Wideband Wideband	Gate / Exp	
Type Freq 7283 0 -12 -24 -36 -48 16 62	0 0.600 250 lk	LookAhead Yes 4k 16k	L - Input - R OUtput Key mode Stereo Monitor Audio Sidechain Passive C1 Compressor / Gate

Setup name: C1 Multimedia Speech 1

The Comp/Exp module is set up as a mid-level compressor to squeeze the top 60 dB of the input dynamic range into the top 40 dB of the output dynamic range, as can be seen from the yellow input/output curve on the graph, This causes a large gain increase at low levels as can be seen from the yellow bar in the Comp/Exp gain reduction meter. The threshold and ratio (between perhaps 1.5 and 3) may be adjusted to vary this compression curve, using the "PeakRef mode" to keep the top of the curve in the right place.

The high gain of low level sounds thus caused (20 dB for the setup illustrated above) may be counteracted by using the Gate/Exp module as a gate to switch off sounds below a certain level. The gate is a conventional one with GateOpen and GateClose thresholds, Attack, Release and Hold adjustments. The values shown are typical of those useful in gating speech, although each case will require different adjustment. For multimedia 8 bit applications, a floor of minus infinity as shown should be used, and the GateOpen thresholds should not be much lower than shown to avoid audible inter-syllable quantization granularity in the final 8 bit sound file.

For gating speech with attack times such as 2 msec or even 5 msec, initial speech sibilants tend to get diminished or cut off unless you use lookahead in the "Yes" position.

Setup name: C1 Multimedia Speech 2

This setup is not illustrated because it is identical to the above except that the Gate button is switched to "Expander" mode (this may be done by clicking on it). This turns the Gate into a downward expander, whose adjustments are similar except that GateClose and Hold are inoperative. The downward expander, used at an extreme "negative polarity" floor setting of -10 N, gives a useful degree of reduction of background noises, and may often be preferable to a gate on account of its gentler "soft knee" reduction of gain.

Equalizer with Wideband dynamics processor

Among the simplest setups to understand are those that combine a single conventional wideband dynamics processor with a simple equalizer, allowing both dynamics processing and EQ to be carried out in a single step. The C1 allows one of its modules to be used for dynamics processing, while the other one is used in a mode where no dynamics processing occurs, but where the use of "split band" plus a suitable "makeup gain" in that band gives equalization.

Setup name: C1 EQ + Gate

The Gate/Comp module in this setup is a normal "wideband" gate, and it may be adjusted in the usual way for gates. It is here operated in "lookahead" mode to minimize loss of starting transients.

The Comp/Exp section is set to ratio 1 so that it does absolutely no processing, but it is in split band mode, which allows the gain of the active frequency band (shown in red on the following graph) to be raised or lowered.



Adjustment of the EQ in no way affects the operations of the gate. The two processes are completely independent.

The equalizer gain in the active band is adjusted by click-and-dragging the Makeup gain Value Window (highlighted in the previous graphic). The filter type, frequency and Q controls may be used to adjust the shape of equalization used. As loaded, the bass below 125 Hz and treble above 4 kHz is boosted (for Makeup gain greater than 0 dB) or cut (for Makeup gain less than 0 dB). The is achieved by using filter type bandreject, which attenuates middle frequencies but lets through low and high.

Other filter type settings obtained by clicking on the type button are:

lowpass, which may be used to boost or cut lower frequencies, highpass, which may be used to boost or cut higher frequencies, or bandpass, which may be used to boost or cut a band of frequencies centered at the preset frequency. The EQ may be switched in or out by clicking on the Comp/Exp bypass button (over the Comp/Exp bar meters).

The Gate may be switched in or out by clicking on the Gate/Exp bypass button (over the Gate/Exp bar meters). Setup name: C1 EQ + LLexpander

This setup combines an equalizer with a conventional wideband low-level (or downward) expander. The equalizer is used as in the C1 EQ + Gate setup, and the low-level expander is implemented by the Gate/Exp module, whose controls are used in the usual way. This setup may be used for example when the equalization brings up background noises, where the expander may help to reduce them again without having the sudden switching effect that can be caused by a gate.

The EQ may be switched in or out by clicking on the Comp/Exp bypass button (over the Comp/Exp bar meters).

The LLexpander may be switched in or out by clicking on the Gate/Exp bypass button (over the Gate/Exp bar meters).

Setup name: C1 EQ + LLcompressor

This setup combines an equalizer (implemented and adjusted as in the previous two examples by the Comp/Exp module makeup gain plus the filter module) and a low-level compressor implemented by the Gate/Exp module.

The Low level compressor brings up the level of low-level sounds below a threshold set by the GateOpen control. The amount of boost, between 0 and +12 dB, is set by adjusting the Floor control. This effect is useful to make inaudible low-level details in a sound more audible, whether these are background ambient noises needed for "atmosphere", quiet from-audience speech at a low level in a conference recording, or quiet noises associated with a louder sound whose increase in level will make the sound seem "hyper-real".

The equalization, adjusted as before, does not affect the low-level compression dynamic processing, but only the tonal quality of the result.

The EQ may be switched in or out by clicking on the Comp/Exp bypass button (over the Comp/Exp bar meters).

The LLcompressor may be switched in or out by clicking on the Gate/Exp bypass button (over the Gate/Exp bar meters).

Setup name: C1 Compressor + EQ

Unlike the previous setups, this has the Comp/Exp module used for wideband dynamic processing and the Gate/ Exp module set up in split mode with the filter as an equalizer.

The equalizer is adjusted exactly as before, except that now the floor control is used to adjust the EQ gain rather than the Makeup control. So the floor control is raised to increase gain in the active band and lowered to decrease it.

The Comp/Exp module is adjusted as a high level compressor in the conventional way with threshold, makeup gain, ratio, attack, release and PDR (program dependent release) controls adjusted in the usual way, and as described in the C1 User's Guide.

The Compressor may be switched in or out by clicking on the Comp/Exp bypass button (over the Comp/Exp bar meters).

The EQ may be switched in or out by clicking on the Gate/Exp bypass button (over the Gate/Exp bar meters).

The Comp/Exp section may also be set up as a mid-level compressor as used in the multimedia speech setups of section 4.1 above.

compressor setups

Here we describe a few more setups combining compression with other functions. For a basic review of compression see sections 3.5 and 3.6 above.

Speech compressor/expander 1

Undo	A : C1 Speech Compress/	Expand 1	A->B Load Save ? WWAVES
Comp/Exp Byp 24 0 18 10 12 20 6 30 0 40 -6 50 -12 60 -12 60 -18 70 -24 80 -24 80 -30 90 -36 100	Peak Ref Expander Makeup Floor 0.0 -co Threshold GateOpen -45.0 -45.0 Ratio GateClose 2.41:1 -45.0 Attack Attack 5.01 \$0.01 Release Release 50 20 PDR Hold 10 0.01 EQ mode EQ mode Sidechain Image: Sidechain	Gate/Exp Byp 12 0 10 10 12 20 24 30 40 -36 40 -48 50 -48 50 -60 60 -72 70 -84 80 -96 90 100 -104.4 -46.9	0db -20 -40 -60 -80 -80 -100 -80 -60 -40 -20 0db
Type Freq 100 0 -12 -24 -36 -48 16 62	0 0.100 ↓ 250 1k	LookAhead Yes 4k 16k	L - Input - R 0.0 0.0 Key mode Stereo V Monitor Audio Sidechain Passive C1 Compressor / Gate

Setup name: C1 Speech Compress/Expand 1

This setup allows a high degree of speech compression and increased level with minimum effect on the natural ness of speech sound. The more natural compressed quality has the downside that the objective unweighted dynamic range can vary more widely than with a wideband compressor.

One important feature is the use of a sidechain EQ that rolls off gently in the bass - note the low Q. (The actual sidechain EQ in dB is 1.5x the number of dB indicated). This reduction of bass gives a much better and more natural compressed sound on speech, and a much less "constricted" sound. Typically the best effect occurs for frequencies from 60 to 180Hz.

The Comp/Exp module is used as a mid-level compressor, and the ratio and threshold controls can be adjusted to taste over a range in which the ratio may be varied between 1.5 and perhaps 5 and the threshold between - 30 and say -70 dB.

Although the sidechain EQ gives a more natural sound, the high degree of compression brings up background noise and reverb. This may be countered by the low-level expander of the Gate/Exp module, set to come into action at the bottom of the compression curve, and also using the sidechain EQ.

This has the effect of reducing noise and also any room reverberation present.

GateOpen is the threshold for downward expansion of noise and reverb, and a setting as high as possible that has little or no audible effect on speech level is recommended, providing that this does not cause too much audible modulation of noise.

One may drag all thresholds (for Comp/Exp and Gate/Exp module) together by click-and-dragging the button between the threshold and GateOpen Value Windows when adjusting threshold to particular speech and background noise characteristics.

The compressor may be bypassed by clicking on the Comp/Exp button, and the expander may be bypassed by clicking on the Gate/Exp button.



Mid-level compression plus speech De-reverberation

Setup name: C1 Compressor + De-Reverb

This setup uses the Comp/Exp module as a mid-level compression unit and squeezes the top 60 dB of dynamic range into 40 dB without compressing the lowest or highest level sounds. Such mid-level compression is useful for making sounds louder and for fitting sounds into the limited dynamic range of say 8 bit multimedia files, without causing the sound to drive peak levels so hard that the sound becomes unpleasant on cheap systems.

Such mid level compression, however, brings up low level parts of the sound by around 20 dB and this can cause unpleasant side effects. The Gate/Exp module can be used in various ways to reduce these side effects.

One side-effect on speech is that room reverberation, which can often already be unpleasant with recordings made outside good studio settings, is further increased to unacceptable levels. The Gate/Exp module here is set up to remove bass signal components at lows levels. This removes background rumble noises and bass components of reverberation tails. Especially on male speech, this has the subjective effect of reducing or removing room reverberation, while having very little effect on speech tonal quality providing the GateOpen threshold is set carefully by ear.

To switch mid level compression in or out, click on the bypass button over the Comp/Exp module bar meters.

To switch speech de-reverberation in or out, click on the bypass button over the Gate/Exp module bar meters.

Controls for mid-level compression:

Comp/Exp module used for mid-level compression.

- Makeup. as needed
- Threshold. -70 to -20.
- Ratio 1 to 3
- Attack, Release, PDR: As needed

Controls for bass de-reverberation:

Gate/Exp module set to bass de-reverberation

- GateOpen threshold. -40 to -15 set by ear to leave signal bass balance unaltered while diminishing reverb bass.
- Attack, Release: As needed usually attack not too short (to avoid distortion of the bass frequencies) and release not too long (to allow quick response to low level in the bass)

Other controls

- Lookahead
- Frequency: In range 200 to 600 Hz for best sound
- Output Level (has same function in this setup as makeup)



Setup name: C1 HLcompress + DeHiss

This is two effects which may be used independently of each other (by bypassing the other) or together. Each may be set-up and optimised independently - adjustments of one do not affect the other.

The Comp/Exp module is set up as a wideband High Level (HL) compressor with Makeup gain. Note that the Comp/Exp ratio is set beyond the "infinite" setting of full limiting to a negative "over-limiting" ratio setting to bring down peaks a little further. This is very effective providing any cancellation nulls lie beyond 0 dB.

The use of compression with gain has, however, the side effect of bringing up any background hiss in a recording. The Gate/Exp module is set up as a de-hisser to minimise the effect of his with least effect on the sound. It is a low-level expander whose action is confined to a carefully-tuned frequency band centered around 7.5 kHz. (See chapter 6 for more information on noise reduction.)

The filters chosen here for de-hissing are generally suitable for hiss with a white, pink or blue spectrum, requiring only perhaps a couple of kHz up or down tuning of frequency - beware however that one is not tuning out hiss emphasis due to a monitor speaker coloration! The tonal effect of the maximum available hiss reduction can be heard by setting the Monitor button to passive mode. The Q setting turns out to be fairly critical - lower will filter out more hiss, but modulation noise - variations of the hiss up and down with the wanted signal, will become

more audible. A Q of 0.5 is about the best compromise, although a Q of up to 0.6 can be used. Generally, the optimum filter settings do not have a lot of leeway and end up close to those in the setup.

The GateOpen threshold setting is the critical one for de-hissing; it should be adjusted up and down until the highest setting that has no significant unwanted effect on tonal quality of the signal is found. This is the optimum de-hiss setting. The setting is a trade-off between loss of treble on the wanted signal and hiss level. On most speech and popular music without very wide dynamics, there is usually a good setting that has very little tonal effect but a subjective hiss reduction of 6 dB or more. The above GateOpen setting was for a white hiss at a level of only -45 dB relative to digital peak - very hissy indeed, but had little tonal effect.

Generally speaking, de-hissing is more difficult on material with a very wide dynamic range, since the tonal effects in quiet passages will be more audible.

Useful Controls:

Comp/Exp module set to high-level compression/limiter.

- Makeup. as needed
- Threshold. -30 to 0.
- Ratio -50 to -5 or 1 to 50 as needed
- Attack, Release, PDR: As needed

Gate/Exp module set to de-hissing

• GateOpen threshold. -25 to -80 set by ear to leave signal treble balance unaltered while diminishing hiss.

Other Useful controls

- Frequency: In range 5000 to 10000 Hz for lowest perceived hiss
- Q: In range 0.45 to 0.6 for best trade off of modulation noise versus hiss reduction
- Output Level (has same function in this setup as makeup)

About Dynamic Noise Reduction

The C1 becomes a very powerful and effective single-ended noise reduction system when loaded with Noise Reduction, De-Hiss or De-Rumble setups. It allows reduction or removal of hiss, rumble or other background noises with the minimum of side-effects.

Noise can usually be reduced or removed from a sound file by heavy filtering of the most audible noise frequencies, but this usually also severely degrades the tonal quality of the wanted signal. Another conventional method of reducing noise is to gate or downward-expand low-level sounds, but this also often has the side-effect of altering or removing wanted low-level sounds that are not in the noise frequency range - and one can usually hear the noise coming and going as signal level alters.

In noise reduction setups, the C1 acts as an intelligent filter that only filters noise when it has to, leaving the wanted signal unfiltered when the wanted signal is loud enough to mask the noise in the frequency band being noise-filtered. It combines the virtues of a noise filter and of a downward expander or gate, while minimising the weaknesses of both.

The principle of dynamic filtering of noise, filtering the signal only when it is below a threshold level in the frequency band of greatest audibility, is not a new one. But in the C1 implementation, the frequency band shape, the nature of the dynamic and sidechain filtering, the attack and decay time constants, the expansion law and the nature of the filter as it dynamically varies have all been optimized for optimum subjective results.

Understanding that the noise reduction is a filter that comes in or fades out as the signal energy varies in the noise band is important. Too much filtering, and the wanted signal is affected. Too little filtering, and a lot of noise remains. The adjustment of the noise reduction setups is to obtain the optimum compromise between too much filtering and too much noise.

But it is also important to be aware of a potential side effect - noise modulation. As the filter comes in or goes out, one may hear also the effect of noise modulation, i.e., the noise level going up and down with the wanted signal. This can be a very distracting effect, and sometimes it is less distracting to leave some noise in than to suffer from excessive noise modulation.

All available forms of noise reduction suffer from audible side effects, and require the user to listen carefully, and to adjust them to minimize these. We believe that the C1 offers the best trade-offs among available dynamic filter noise reducers, having fewer side effects for a given degree of noise reduction.

The use of the C1 noise reduction is now described with reference to the setups provided in the C1 setups library. You should load the setups described and try them out on appropriately noisy soundfiles, adjusting them as described below.

C1 Hiss Reducers



Setup name : C1 Noise Reducer

This is intended for Hiss reduction. It provides two stages of processing. The dynamic noise filtering is provided by the Gate/Exp module, used in Bandsplit mode with the filtering shown on the frequency response graph. The band shown in red is the band whose level is reduced in quiet passages to reduce hiss.

An optional wideband low-level expander is provided by the "Comp/Exp" module. This can, optionally, be used to gently pull down further the level of any very quiet passages in which the reduced noise may still be audible. It can be switched in or out at the user's option by clicking on the Comp/Exp button to bypass the Comp/Exp module. The noise reduction works very effectively on its own, and this low-level expander is provided simply to provide a further reduction in quiet passages if desired.

The adjustments to the Noise Reducer (Gate/Exp module) are essentially just three controls:

(i) GateOpen threshold in the Gate/Exp module.

Moving this upward will increase the amount of filtering both of the signal and noise, moving it down will decrease the amount of filtering. Normally, one would use the highest setting that does not audibly degrade the wanted signal or cause objectionable modulation noise. The precise setting is a judgement depending on the skill and aims of the user.

Hint. It is often best to adjust the GateOpen level by click-and-dragging the button (highlighted in the previous picture) between the threshold and GateOpen Value Windows. This has the effect of moving both GateOpen and Comp/Exp thresholds up and down together. The reason for this is that if you decide to use the low-level expansion option for further reduction of noise in quiet passages, then the level at which this is effective remains matched to the noise reduction GateOpen threshold.

(ii) Filter frequency.

This adjusts the center frequency of the frequency band subjected to filtering. Normally, this would lie somewhere between 5 and 10 kHz depending on the tonal balance of the hiss, with 7.5 kHz being a typical frequency for "white" hiss noise.

(iii) Filter Q.

This adjusts the width of the filtered frequency band. The lower the Q, the wider the band and greater the amount of perceived noise reduction, but lower Q's also increase the risk of audible modulation noise effects. In practice a Q of between 0.5 and 0.6 is usually best for this reason.

Hint: When adjusting the filtering, you may like to switch the Monitor button to Passive mode. This monitors the "passive" band (shown on the frequency response graph in blue), which corresponds to the maximally filtered signal. This allows you to hear the effect of the filtering on its own without the dynamic processing, and is useful for tuning for minimum hiss audibility.

Switching between "Passive" and "Audio" monitor mode, by clicking on the Monitor button also provides a direct comparison between conventional filtering and the noise reduced audio. In many cases, it will be found that the noise reducer mode, once adjusted, almost magically gives the same noise reduction as the Passive mode, but with little of the treble loss given by simple filtering.

Other controls that can be used.

Normally the above controls are all you need ever use for noise reduction adjustment, but a finer control is possible if one has extra patience. The Floor control can be moved from its preset -10 N setting to less extreme "negative polarity" settings between -10 N and -100 N or to floor settings below say -6 dB. This gives a reduced noise reduction effect, but also reduces modulation noise. A combination of floor and GateOpen threshold may give better results than GateOpen threshold alone, especially where only a partial degree of noise reduction is required and where the noise background fluctuates in level. However, juggling the effect of these two controls is more difficult than just using GateOpen threshold.

When the wideband expander is used, its effect may be adjusted by operating its threshold independently of that of GateOpen, although the preset 30 dB difference in values is generally roughly right. The Comp/Exp module's attack, release and PDR values may be adjusted in the usual way for an expander for optimum sound.

In general, you should usually not alter the attack and release time constants from their preset values in the Gate/ Exp module - the values used are about optimum for hiss filtering applications, and also quite well matched to high frequency "crackle" and 78 rpm surface noise. The use of high-pass filter mode for hiss filtering is generally less effective than the bandpass filtering shown above, mainly because it gives more audible hiss noise for a given degree of tonal alteration, being less well matched to the ear's sensitivity to hiss.. However, if it is used, generally use Q = 0.6 to minimize modulation noise effects.

This kind of setup can also be effective with other kinds of high frequency noise, including "scratch" noise from old 78 rpm recordings, and even certain kinds of electrical buzz. For 78 rpm noises, the noise reduction may work better if the Gate/Exp Attack is increased to 5 msec, with the Gate/Exp Release left at 1 msec.

Setup name: C1 Noise Reducer 2

This is a very powerful noise reducer also designed to reduce hiss and other high frequency noises. The Gate/Exp module is used as a noise reducer identical to that in the above noise reducer. The Comp/Exp section, however, is configured as a second bandsplit low-level expander also capable of achieving noise reduction. The two sections combined give twice the amount of expansion action at low levels. This setup is useful where the maximum degree of noise reduction is required without too much tonal effect on the wanted signal.

The adjustments to C1 Noise Reducer 2 are essentially just three controls:

(i) Click-and-drag on the highlighted button between threshold and GateOpen.

Moving this upward will increase the amount of filtering both of the signal and noise, moving it down will decrease the amount of filtering. Normally, one would use the highest setting that does not audibly degrade the wanted signal or cause objectionable modulation noise. The precise setting depends on the skill and aims of the user.

(ii) Filter frequency.

This adjusts the center frequency of the frequency band subjected to filtering. Normally, this would lie somewhere between 5 and 10 kHz depending on the tonal balance of the hiss, with 7.5 kHz being a typical frequency for "white" hiss noise.

(iii) Filter Q.

This adjusts the width of the filtered frequency band. The lower the Q, the wider the band and greater the amount of perceived noise reduction, but lower Q's also increase the risk of audible modulation noise effects. In practice a Q setting of between 0.5 and 0.6 is usually best for this reason.

Hint: When adjusting the filtering, you may like to switch the Monitor button to Passive mode. This monitors the "passive" band (shown on the frequency response graph in green), which corresponds to the maximally filtered signal. This allows you to hear the effect of the filtering on its own without the dynamic processing, and is useful for tuning for minimum hiss audibility.

Switching between Passive and Audio monitor mode, by clicking on the Monitor button also provides a direct comparison between conventional filtering and the noise reduced audio.

Longer attack times of 5 ms may be used in the two modules - this will have the effect of making the noise reducer 2 work better with impulsive noises such as those on 78 rpm records. Leave the release time at 1 msec.

Either module may be switched out by clicking the Comp/Exp or Gate/Exp button to bypass the relevant module.

Setup name: C1 Gate Noise Reducer

This noise reducer uses a bandsplit gate rather than an expander to cut out noise. This is extremely effective in quiet passages, but results in audible sudden appearance and disappearance of hiss. This can be acceptable when one sound source (e.g. a noisy keyboard or electric guitar) is part of a mix. This setup, also includes, in the Comp/Exp module, a second gentle low-level bandsplit downward expander to provide some noise reduction above the gate threshold, so as to make gate switching effects on noise less audible.

The adjustments for the Gate Noise reducer are the same as for Noise Reducer 2 above.

In addition, The GateClose (normally set between 0 and 9 dB lower than GateOpen) and Hold controls of the gate can be used in the usual way to keep the gate open longer (during which it will let through more noise) if desired to preserve more treble of the wanted sound.

Rumble Reduction

Setup name: C1 Rumble Reducer

This setup reduces bass rumble noises in much the same way that the previous setup reduced hiss noises. Many recordings have excessive low frequency or "rumble" noise, either due to traffic, air conditioning or similar sound sources, or due to air currents or other spurious noises transmitted to the microphone body. Rather than filter out all low frequency sounds to remove these, a rumble reducer that retains bass response for higher level sounds is often preferable, and can have virtually inaudible action.

Its mode of operation is identical to the previous noise reducer with just two exceptions: the filters are now tuned for removing bass frequencies rather than treble, and (ii) the Gate/Exp Attack and Release time constants are set to be longer to minimize bass distortion.

The filter frequency control may be adjusted by ear with the Monitor button in Passive mode to minimize the audible rumble noise, Clicking on the Monitor button back to Audio mode, the GateOpen threshold may then be adjusted to reduce rumble level while having the minimum effect on the tonal quality of wanted sounds.

As before, the Comp/Exp section is set up as a gentle low-level expander to reduce any residual noise in quiet passages, and it may be adjusted by moving the threshold up or down. It may be removed from the algorithm by clicking the Comp/Exp button into bypass mode.

Hint: If the rumble level is very high, it may not be possible to remove rumble without some tonal effect on the wanted signal. In this case, one should first process the sound file to remove the very lowest rumble frequencies by using the Waves Q10 parametric equalizer with a steep cut high-pass filter, for example one taken from the Q10 setups Library bandlimit filters such as the high-pass filter from the Broadcast44.1.setup, adjusted to cut off
at the highest acceptable frequency. In the case of deep rumble, this will remove the largely inaudible but very high energy very low frequencies. The C1 Rumble reducer should then be used on the processed file to reduce the more audible remaining components of rumble.

If a Q10 parametric equalizer is not available, the C1 itself may be used as a high pass-filter for the first stage of processing by loading the setup C1 high-pass filter described below. This may be adjusted in frequency and used for processing to remove the lowest rumble frequencies before the C1 Rumble reducer is applied. However, the sharpness of filtering is less good than can be achieved using the Q10.

Setup name: C1 High-pass filter.

This uses the "Active" monitor mode position (normally used for monitoring the sidechain signal) as a 36 dB per octave high pass filter for applications such as cutting out infrasonic and deep bass rumble sounds. The cut-off frequency may be adjusted by adjusting the frequency Value Window, which actually shows the frequency at which the filter is 9 dB down. An octave higher the filter is only about 1.5 dB down, and an octave lower about 30 dB down. This filter may be used as the first stage in processing where the C1 Rumble Reducer described above is used as the second stage of processing.

Removal of miscellaneous noise

The reduction of noise or background sounds with arbitrary frequency spectrum, rather than just hiss or rumble, involves a greater degree of user patience, experimentation and skill in setting up the C1. The method is as follows.

Load either the C1 Noise Reducer or the C1 Rumble Reducer setup. Initially, click the Monitor button to Passive mode, and tune the filter, using the type, frequency and Q controls for a useful degree of reduction of the unwanted background noises, while making sure that the removal of wanted sounds is not excessively severe. (It is in making this judgement that personal skill and experience are important). Then click the Monitor button back to Audio mode.

One should set up the Attack and Release times of the Gate/Exp section according to the active band response shown in red on the frequency response display. The rule here is to make Gate/Exp Attack and Release times roughly equal, or Release perhaps only twice as large as Attack, and to set Attack at around 1 ms if the active band response shown in red on the frequency response display is shown as below say -15 dB at all frequencies below 1 kHz. If the active band response is more extended at low frequencies, then longer Gate/Exp attack and release times should be used to prevent nonlinear distortion, with Attack and Release times being increased inversely as the bass response frequency of the active band is lowered, up to around 7 ms for an extended bass component in the active band.

One then sets the GateOpen threshold to achieve the desired trade-off between noise reduction and unwanted effects on the wanted signal. In the event that a satisfactory degree of noise reduction is obtained only at the expense of excessive low-level "pumping" of the signal, there are two things you can do to improve the trade-off. First, you can reset the filter so that the active band affects a smaller frequency range, affecting less of the wanted signal. If this does not produce the desired results, you can use the floor control set to somewhere in the range -5 dB to -15 dB to reduce the amount of wanted-signal pumping. This will also reduce the degree of noise reduction, but may give a more satisfactory trade-off, achieving moderate noise reduction but with fewer side effects.

In some situations, you may prefer to set the Gate/Exp module mode to "Gate" rather than "Expander", especially if the noise is only obtrusive in "quiet" gaps in the program material and not during loud passages, and generally in that application you may use somewhat longer Gate/Exp release times of up to say 100 msec. Such gate settings may be used also if the sound is just one component in a mix.

Chapter 7 - De-Essers

On De-Essers

De-essers are essentially dynamic processors that reduce the signal gain during peak-level sibilants ("esses") by operating a gain reduction during high levels in the "Ess" frequency band typically centered at about 7 kHz. They are used to reduce excessive sibilance in speech and other vocal sounds.

Two different kinds of de-essers can be implemented on the C1:

Sidechain de-essers, which reduce the gain of all frequencies in the signal during a high level in the "Ess" frequency band. This is the most familiar type of de-esser, and

Bandsplit de-essers, which reduce the gain only in the "Ess" frequency band but leave other frequencies unaltered.

De-essers also differ in the degree of gain reduction as the "Ess" level increases, and different implementations produce different tonal qualities to de-essing. Which is preferred is partly a question of subjective choice, and partly a question of what other functionality (e.g. compression, gating, noise reduction, etc.) is also available with a particular de-esser in the C1.

Which kind of de-esser?

There are 5 different de-esser modes possible on the C1. These are:

- (1) using the Comp/Exp module in sidechain mode
- (2) using the Comp/Exp module in split mode
- (3) Using the Gate/Exp module in Expander sidechain mode + gain makeup
- (4) Using the Gate/Exp module in Expander split mode + Comp/Exp makeup in split mode .
- (5) Using the Gate/Exp module in Gate split mode + Comp/Exp makeup in split mode.

Each of these de-essers have a different sound and also allow use with different combinations of processing. So the choice will be made partly on sound and partly on other processing one wishes to do at the same time.

Sidechain v. bandsplit de-essers

Sidechain de-essers have the least tonal effect on the sound, their downside being that their gain variations can sometimes be heard as gain pumping or instability.

Bandsplit de-essers work by varying the gain in the Ess-band. Generally, these have far fewer gain instability effects, since they only affect the Ess band, but their downside is that they can also cause a tonal dulling. This effect tends to be least if the Gate-Split de-esser is used and most if the Comp/Exp module split de-esser is used.

The setups below all use one dynamics module for the main de-essing and the other for other tasks. Where stated, the other module may be switched out if not required.

Compressor + De-Esser

Setup name: C1 Compressor + DeEsser 1

This setup combines a compressor (here set to mid-level compression using the Comp/Exp module in PeakRef & wideband mode) and a de-esser implemented using the Gate/Exp module in sidechain expander mode. The -12 dB output gain is intended to compensate for the +12 dB floor gain used in the de-essing in the Gate/Exp module. Any additional desired change of output gain may be used.

The compressor may be switched in or out by clicking the bypass button above the Comp/Exp module meter bars. The de-esser may be switched in or out by clicking the Gate/Exp bypass button above the Gate/Exp meter bars.

De-Essing adjustments

The GateOpen level is used to adjust the degree of de-essing. You can see by the grey area at the top of the gain reduction meter of the Gate/Exp module, and by the level metering on the light blue line on the input/output graph how the high-level sibilant sounds are being reduced in level. The frequency band of de-essing, shown in red on the frequency response graph, may be adjusted by altering the filter frequency (typically in the range 5000 to 11000 Hz) and the Q (typically 0.4 to 0.6).

Compressor adjustments

The ratio (typically in the range 1.0 to 3.0) of the compressor, and its threshold (typically in the range -35 to -60 dB) may be used to adjust the compression effect to taste, and the attack, release and PDR may also be altered as usual. The degree of low-level gain increase given by the compressor is indicated by the yellow bar of the gain reduction meter of the Comp/Exp module.

De-Ess & De-Hiss

Setup name: C1 De-Ess + De-Hiss 1

This setup uses the Comp/Exp module in sidechain mode as a de-esser, and the Gate/Exp module in split expander mode as a de-Hisser, They share the filter setting, but are otherwise independent. Either module may be bypassed or substituted by another setup for that module without affecting the functioning of the other.

The de-esser may be switched in or out by clicking the bypass button above the Comp/Exp module meter bars. The de-hisser may be switched in or out by clicking the Gate/Exp bypass button above the Gate/Exp meter bars.

De-essing adjustments

The Comp/Exp module is set up to "duck" in level whenever high levels arrive in the active "Ess" frequency band. Altering the threshold control alters the degree of de-essing. You can see by the level metering on the yellow line on the input/output graph and the gain reduction meter (shown in color as a red bar) in the Comp/Exp module how the high-level sibilant sounds are being reduced in level. The ratio control in the range say 2 through 50 to - 5 can also be used to alter the quality of the de-essed sound. Generally, ratio settings near 2 or 5 give a moderate degree of de-essing but few unnatural side effects. Ratios near 20 to 50 or -50 to -25 give more effective de-essing, but more side effects, and negative ratios if overdone with too low a threshold can result in an over-processed effect that sounds like the speaker has lost his/her teeth!

De-hissing adjustments

The Gate/Exp module is set up as a de-hisser, and the degree of de-hissing can be adjusted by moving the GateOpen threshold up or down until the effect on tonal quality is minimal. This allows a trade-off between degree of hiss reduction and alteration of tone quality. You can see from the gain reduction meter (shown in color as a red bar) in the Gate/Exp module how much hiss reduction is taking place.

Small adjustments in frequency and Q can be used to optimise either the de-hissing or de-essing, or some compromise between the two.

Hint. When using both effects together, in setting up the de-hissing or de-essing, it is helpful to switch the other processing module to bypass by means of the bypass button above the bar-meter section of the other module. That way one can avoid confusion caused by the interaction of the two processes. Once each is set up individually, switch in both and make final adjustments to each to minimise any tonal dulling. Note that to avoid dulling, the threshold should always be set at least 15 dB, preferably more, below GateOpen.

De-Ess + LowLevel compressor

Setup name: C1 De-Ess + LLcompress

This setup uses the Comp/Exp module in sidechain mode as a de-esser and the Gate/Exp module in wideband expander mode with positive floor gain as a low-level compressor to make sounds louder.

The de-esser may be switched in or out by clicking the bypass button above the Comp/Exp module meter bars. The low-level compressor may be switched in or out by clicking the Gate/Exp bypass button above the Gate/Exp meter bars.

The Comp/Exp module is set up to "duck" in level whenever high levels arrive in the active band. Altering the threshold control alters the degree of de-essing. The ratio control in the range say 2 through 50 to -5 can also be used to alter the quality of the de-essed sound.

The filter may be tuned in frequency and Q slightly to alter the predominant de-essing frequency band. One can monitor the selection of Ess frequencies by switching the monitor button to "sidechain".

Control of low-level compression

The floor control, between 0 and 12 dB and the GateOpen threshold of the expander module can be used to adjust the degree of low-level compression. Higher settings of floor give a higher degree of compression. The threshold adjusts the sound level below which compression occurs. The attack and release times of the low-level compressor can be adjusted to taste in the usual way for a compressor. Use of lookahead in the "yes" position is usually recommended.

Bands	olit De-Ess.	De-Hiss and EQ	

Setup name: C1 De-Ess + EQ + De-Hiss 2

This is very similar to C1 De-Ess + De-Hiss 1 except that now the de-esser is working in bandsplit rather than sidechain mode. Adjustments are made in much the same way as for the C1 De-Ess + De-Hiss 1 setup, although the sound quality of this de-esser is rather different, often requiring significantly lower threshold settings.

A second difference of this setup is that it can additionally be used to alter the EQ by altering the level of the active band, using the makeup gain to boost or cut the active band. This helps to compensate for any slight dulling of mid-level sounds given by the combination of de-essing and de-hissing, or may be used to provide additional tonal alteration.

Bandsplit De-Hiss, De-Ess and EQ

Setup name: C1 De-Hiss + EQ + De-Ess 3

This setup combines a de-hisser using the Comp/Exp module in PeakRef & bandsplit mode) and a de-esser implemented using the Gate/Exp module in bandsplit expander mode. The -12 dB makeup gain in the Comp/Exp module is intended to compensate for the +12 dB floor gain used in the de-essing in the Gate/Exp module. Any additional desired change of output gain may be used.

The de-hisser may be switched in or out only by altering the value of ratio to 1, since its make-up gain is part of the functioning of the de-esser. The de-esser may be switched out by clicking the Gate/Exp bypass button above the Gate/Exp meter bars and also altering the makeup gain to 0 dB.

De-Essing adjustments

The GateOpen level is used to adjust the degree of de-essing. You can see by the grey area at the top of the gain reduction meter of the Gate/Exp module, and by the level metering on the light blue line on the input/output graph how the high-level sibilant sounds are being reduced in level. The frequency band of de-essing, shown in red on the frequency response graph, may be adjusted by altering the filter frequency (typically in the range 5000 to 11000 Hz) and the Q (typically 0.4 to 0.6).

De-Hisser adjustments

The threshold (typically in the range -65 to -100 dB) may be used to adjust the de-Hissing effect to obtain the best compromise between hiss level and alteration of tonal quality of the wanted sound. The degree of hiss reduction at each moment is indicated by the red bar of the gain reduction meter of the Comp/Exp module.

You should try and ensure that the de-hissing threshold is not too close to the de-essing threshold, since otherwise there will be a significant loss of the "ess" frequencies at all signal levels.

EQ adjustments.

You can alter the overall tonal quality further by adding additional EQ in the active band. This may be done by varying the Makeup gain up or down around its normal level. This can also help to compensate for tonal losses due to the combined effect of the de-hissing and de-essing.

Setup name: C1 De-Hiss + EQ + De-Ess 4

This is used identically to the previous set-up, differing from it only in that the Gate/Exp module is now in gate rather than expander mode.

However, the more extreme sudden change of gain above the "Ess" threshold given by the gate means that the subjective effect of this setup is very different. In some cases, it will be found to be very effective at de-essing, but in other cases, the effect may be found to be overdone.

Chapter 8 - Enhancers

About enhancement

Quite often one wishes to add clarity and/or depth to a soundfile to give it more "pizazz", intelligibility or impact. This can be done using conventional equalization, but all too often this has the side effects of making bass boomy or treble harsh and uncomfortable. The C1 enhancement modes allow clarity and depth to be added either to a single sound or to a complete mix with far fewer unpleasant side effects.

The general idea of these enhancers is to boost bass or treble frequencies only at lower levels below a threshold, but not at high levels. This ensures that the loudest, and potentially most oppressive, parts of the sound do not have an exaggerated bass or treble, while the quieter parts have the extra warmth or sparkle obtained by being boosted in level. Generally, the choice of boosted frequencies is fairly critical for optimum effect, although the choice may depend to some extent on the sound being processed.

Speech and vocal enhancer (with optional mid-level compressor).



Setup name: C1 Speech Enhancer

The Gate/Exp module is here used in bandsplit mode to boost speech presence frequencies for enhanced intelligibility without the degree of unpleasant harshness or sibilance associated with conventional presence EQ. These frequencies are boosted by the setting of the floor control at low levels only, the threshold for those levels being set by the GateOpen threshold control.

The selective boosting only of lower levels in the presence band achieves a clarity and an ability for speech or vocals to cut through noisy listening environments (or dense mixes) without the unpleasantness given by simple EQ used to the same end, and the results still sound reasonable on high-quality reproduction systems.

Adjustment

The degree and subtlety of the effect may be varied by adjustment of the floor level boost between 0 and +12 dB (which acts as a presence gain control), with higher settings giving a greater effect, and the GateOpen threshold - lower GateOpen settings of threshold give a more subtle effect and higher ones a more blatant effect.

This is an effect that can be used subtly, almost subliminally, to just make speech (or musical vocals) seem more "present" without any very obvious change of tonal quality in a mix, or it can be used to provide blatant tonal changes with fewer than usual side effects.

This is usually best with frequency and Q settings roughly as shown - lower frequencies tends to start emphasising harshness, and higher frequencies emphasise brightness rather than vocal intelligibility - however, this may be an effect that may be wanted to brighten mixes without serious side effects.

The Comp/Exp module, shown bypassed above, is set up for a moderate degree of wideband mid-level compression, suitable for making speech louder without making peak levels drive the reproduction system too hard too much of the time. The compression effect may be adjusted by moving the threshold up or down and the compression ratio between say 1 and 3.

Brightness enhancer + compensating noise control.



Setup name: C1 Treble Enhancer

The primary function of this setup is to act as an effective "brightness enhancer". This is provided by the Gate/ Exp module in split mode. It will be noted that many consumers prefer the sound of Dolby^{TM} B cassettes without Dolby^{TM} decoding because of the low-level brightness enhancement done by that process, and the basic enhancement idea here is similar, but more controllable. High frequencies are boosted by up to 9 dB, the degree of boost being controlled by the floor control, and this boost takes place below a threshold set by the GateOpen threshold. These two controls may be tuned - floor between 0 and + 12 dB and GateOpen typically between -20 and -60 dB, for optimum enhancement of a particular mix or sound. The effect gets more subtle as GateOpen and Floor are lowered, and more blatant as they are raised.

The yellow bar on the Gate/Exp gain reduction meter can be used to monitor the degree of gain increase in the treble moment by moment.

The Comp/Exp module is here used in an unusual low-level expansion bandsplit mode to solve the problem that the enhancer also raises the level of any low-level hiss noise present, by acting as a gentle noise reducer at very low levels to pull the noise back down again. Generally speaking, the threshold should be set very low so as to ensure

that the noise reduction does not affect the mid-levels that are being boosted by the enhancer. The threshold should be adjusted by ear to ensure that the noise reduction is effective without affecting enhancement effects too much.

The red bar on the Comp/Exp gain reduction meter can be used to monitor the amount of low-level treble noise reduction moment by moment.

There is nothing sacred about the filter frequency and Q settings used in this setup - but it is often found that the high-pass setting works best at maximum Q of 0.6. The setting shown is one that generally increases brightness without too much mid range harshness creeping in. You may also care to experiment using settings with bandpass filter type tuned with frequency in the region 3 to 9 kHz and having Q's in the region 0.3 to 0.6.

Bass enhancer + compensating rumble control.

Setup name: C1 Bass Enhancer

This is the bass equivalent of the brightness enhancer above.

The primary function of this setup is to act as an effective "bass enhancer". This is provided by the Gate/Exp module in split mode. It provides an added bass richness or depth, even played back via modest equipment, without making the bass overwhelming and unpleasant. Selected low frequencies are boosted by up to 9 dB, controlled by the floor control, and this boost takes place below a threshold set by the GateOpen threshold. These two controls may be tuned - floor between 0 and + 12 dB and GateOpen typically between -15 and -50 dB, for optimum enhancement of a particular mix or sound. The effect gets more subtle as GateOpen and Floor are lowered, and more blatant as they are raised.

The yellow bar on the Gate/Exp gain reduction meter can be used to monitor the degree of gain increase in the bass moment by moment.

The Comp/Exp module is here used in an unusual low-level expansion bandsplit mode to solve the problem that the enhancer also raises the level of any low-level rumble noise present, by acting as a gentle noise reducer at very low levels to pull the noise back down again. Generally speaking, the threshold should be set very low so as to ensure that the noise reduction does not affect the mid-levels that are being boosted by the enhancer. The threshold should be adjusted by ear to ensure that the noise reduction is effective without affecting enhancement effects too much.

There is nothing sacred about the filter settings used in this setup - but it is often found that the bandpass filter type setting with frequency tuned to predominant frequencies in the range 100 to 150 Hz works best at maximum Q of 0.6. The setting shown is one that generally increases the bass in contemporary popular music without too much lower mid range muddiness creeping in. You may also care to experiment using lowpass filter type setting with frequency tuned in the frequency region up to 350 Hz and having Q's of 0.6.

Generally, the lower the bass frequencies selected, the longer should be the Gate/Exp release times to minimise possible bass distortion effects.

Bass & Treble enhancer + compensating noise control.

Setup name: C1 Bass/Treble Enhancer

This is the equivalent of the two enhancers above when enhancement of both frequency extremes at the same time is required.

The primary function of this setup is to act as an effective "bass and treble enhancer". This is provided by the Gate/Exp module in split mode. It provides an added bass richness or depth and an added brightness, even played back via modest equipment, without making the bass or treble overwhelming and unpleasant. Selected low and high frequencies are boosted by up to 9 dB, with boost controlled by the floor control, and this boost takes place below a threshold set by the GateOpen threshold. These two controls may be tuned - floor between 0 and + 12 dB and GateOpen between -15 and -50 dB, for optimum enhancement of a particular mix or sound. The effect gets more subtle as GateOpen and Floor are lowered, and more blatant as they are raised.

The yellow bar on the Gate/Exp gain reduction meter can be used to monitor the degree of gain increase in the bass and treble moment by moment.

The Comp/Exp module is here used in an unusual low-level expansion bandsplit mode to solve the problem that the enhancer also raises the level of any low-level rumble and hiss noise present, by acting as a gentle noise reducer at very low levels to pull the noise back down again. Generally speaking, the threshold should be set very low so as to ensure that the noise reduction does not affect the mid-levels that are being boosted by the enhancer. The threshold should be adjusted by ear to ensure that the noise reduction is effective without affecting enhancement effects too much.

There is nothing sacred about the filter settings used in this setup - but it is often found that the band-reject filter type setting with frequency tuned to predominant frequencies in the range 500 to 1000 Hz works best at Q between 0.15 and 0.3 so as to give the desired bass and treble crossover frequencies on the graph. The setting shown with crossover frequencies around 125 Hz and 4 kHz is one that generally increases the bass and brightness in contemporary popular music without too much lower mid range muddiness or upper mid range harshness creeping in.

Cheap System enhancer + compensating noise control.

Setup name: C1 CheapSystem Enhancer

This is a variation of the Bass & Treble enhancer above, designed to solve the problem of an enhanced sound designed to be played on cheap crummy systems, such as many multimedia systems, AM radios, cheap portable headphone systems or even utilitarian PA and muzak-type systems, preferably while still sounding good on high-end systems. It actually manages to incorporate three functions: a high level bandwidth reducer, a low level bass-and-treble enhancer, and a single ended noise reducer.

Cheap systems often have limited frequency range at the two frequency extremes, and they often distort if one attempts to push much level into them at the frequency extremes, so that limitation of bandwidth at high input level is necessary to get the best out of such systems - but this often makes the sound terrible over better systems.

This setup combines bandwidth limitation between 250 and 4000 Hz at high levels - reducing levels at the frequency extremes by 6 dB, with 12 dB bass and treble enhancement at low levels. This subjectively causes an overall increase in bass and treble - even on cheap systems, with a nice-sounding enhancement, while actually limiting the degree to which cheap systems are exposed to signals they cannot handle. The result is that cheap systems can be played louder without distortion and yet give enhanced bass and treble at the same time. The effect is sufficiently subtle that the enhancement also sounds good on high-quality systems.

The main control here is the GateOpen threshold control, which should be set somewhere between -15 and -35 dB on typical programme material. The gain Makeup control sets the degree of high-level attenuation of bass and treble, which should always be less than the degree of boost of floor, which sets the amount of lower-level enhancement. The Comp/Exp threshold control, which should be set low by ear to have minimum side effects on the wanted signal, acts as a low-level noise reducer for any noise present in the source material.

The yellow bar on the Gate/Exp gain reduction meter can be used to monitor the degree of gain increase in the bass and treble moment by moment, but the degree of gain increase in this setup is reduced by the makeup value, in the case shown above, by 6 dB.

Compressors + Enhancers

These 4 setups use the Comp/Exp module in wideband mode for compression and the Gate/Exp module in bandsplit mode for enhancement. The two modules operate effectively independently on one another. These setups are particularly useful for maximising impact of sound files that have to be squeezef into a narrow dynamic range, such as for multimedia or broadcast material.

Setup names:

- C1 HLcompress+bass/treb enhance
- C1 HLcompress+treble enhance
- C1 MLcompress+bass/treb enhance
- C1 MLcompress+treble enhance

The Gate/Exp and Filter modules of these setups are the same, and used in the same way, as those for the C1 bass/ treble enhancer or the C1 treble enhancer above.

The Comp/Exp module of these setups is used as a compressor. The HLcompress setups are high-level compressors that may be used conventionally, similarly to as describedefor the C1 HLcompress + De-Hiss setup. The MLcompress setups are mid-level compressors similar to, and used in the same way as, the Comp/Exp sections of Chapter 4: C1 Multimedia Speech 1, C1 Multimedia Speech 2 and Chapter 5: C1 Compressor + De-Reverb.

Enhancers + Gate

These two setups provide enhancement plus gate, using the Comp/Exp with Filter modules as an enhancer and the Gate/Exp module in Wideband Gate mode as a gate.

Setup name: C1 Treble enhance + Gate

The Comp/Exp module in split mode of this setup acts as an effective "brightness enhancer". Low-level high frequencies are boosted by up to 10 dB, the degree of boost being controlled by the ratio control, and this boost takes place below a threshold set by the threshold. These two controls may be tuned - ratio between 1 and 1.60 and threshold typically between -40 and -80 dB, for optimum enhancement of a particular mix or sound. The effect gets more subtle as threshold and ratio are lowered, and more blatant as they are raised.

The yellow bar on the Comp/Exp gain reduction meter can be used to monitor the degree of gain increase in the treble moment by moment.

There is nothing sacred about the filter frequency and Q settings used in this setup - but it is often found that the high-pass setting works best at maximum Q of 0.6. The setting shown is one that generally increases brightness without too much mid range harshness creeping in. You may also care to experiment using settings with bandpass filter type tuned with frequency in the region 3 to 9 kHz and having Q's in the region 0.3 to 0.6.

The Gate is a conventional fully-functioned wideband gate. The GateClose threshold is preset 3 dB lower than the GateOpen threshold so as to reduce gate "chatter", and the hold control ensures a minimum gate on time which again helps prevent chatter. Attack and Release may be adjusted in the usual way. If it is desired to retain some of the background atmosphere during quiet passages, the floor control may be used to adjust the remaining signal level when the gate is "off".

Setup name: C1 Bass/treb enhance+Gate

The Comp/Exp module in split mode of this setup acts as an effective "enhancer" of ther bas and treble frequency extremes. Low-level low and high frequencies are boosted by up to 10 dB, the degree of boost being controlled by the ratio control, and this boost takes place below a threshold set by the threshold. These two controls may be tuned - ratio between 1 and 1.60 and threshold typically between -40 and -80 dB, for optimum enhancement of a particular mix or sound. The effect gets more subtle as threshold and ratio are lowered, and more blatant as they are raised.

The yellow bar on the Comp/Exp gain reduction meter can be used to monitor the degree of gain increase in the bass and treble moment by moment.

There is nothing sacred about the filter frequency and Q settings used in this setup - and the bass and treble ranges that are enhanced may be adjusted by varying the Q and frequency around the preset values.

The gate is adjusted in the same way as in the previous setup.

Chapter 9 - Keying Setups

About keying

These setups make use of the key mode button, and allow a signal in one of the two channels to control the dynamics of the other.

The most common uses of this kind of facility is ducking and gating. Ducking reduces the gain of one signal when the other is above a threshold. It is normally used to make a background sound quieter when a foreground sound must be heard over the background. It is commonly used with a speech signal to control the dynamics of loud background noises such as audience applause during a broadcast announcement or commentary, so that the speech is clearly audible over the background.

The other common use is keyed or triggered gating, where the level in one track is used to operate a gate in a second track, so that sounds in the second track are allowed through only rhythmically in time with the occurrence of sounds in the first track. For example, one can use a drumbeat on the left track to control the gating of a steady sound in the right track so as to create a different percussion sound having the tonal quality of the steady sound, in time with the original drumbeat. By using sidechain filtering, one can pick out particular frequency components of the drumbeat such as a snare, bass drum or cymbal sound to key or trigger the other sound.

By using bandsplit processing with ducking or triggered gating, one can process just a selected frequency band rather than the full signal spectrum.

Keying can also be used with expanders instead of gates, which has particular uses in creating rhythmic tracks out of steady sounds such as natural steady sound effects like wind noises, rainfall, babbling streams, waterfalls, crowd noise, audience applause etc., as well as artificial continuous sounds like looped sounds, synthesiser tracks etc. If used with bandsplit, this can give otherwise steady sounds that vary in quality rhythmically in time with the original keying track. A sound effect can be made to take on musical rhythmic qualities by being transformed by a keying effect. This leads to a considerable variety of creative musical effects in dance and ambient musics, as well as ways of integrating sound effects and music in radio, TV, video and film drama, documentary, games and advertising applications. The keyed EQ expansion effect described below is particularly flexible and effective in these applications.

Preparing and processing files for keyed effects

Because Plug-Ins for Sound Designer IITM are stereo processors capable of taking in and processing only two channels of sound at a time, to obtain the keying effect, it is first necessary to prepare a sound file having the keying sound that controls the dynamics on one channel, and the sound that is to be dynamically processed - the keyed or triggered sound on the other. The C1 provides the option for either channel to be allocated to the keyed or keying sound. For ease of description, we assume that the keying or triggering sound is in the left channel and that the sound to be dynamically processed or keyed is in the right channel. For this option, the Key Mode button is in the L -> R mode. All keying setups are loaded in this mode.

When processing sound files in this mode, only the dynamically processed sound on the right channel will be present, with nothing on the left channel. If you wish to preserve the original keying sound as well on the left

channel, in an unprocessed form, you should before processing select the right channel signal only. Processing will then result in a stereo file with the unprocessed keying signal on the left, and the processed keyed signal on the right channel. This is useful in cases such as ducking where it is usually desired to mix these two signals together at a later stage.

Hint: If you wish to process or key a stereo signal by a mono keying signal, this cannot be done directly on the C1 because of its limitation to two channels of input. However, a stereo workaround exists as follows. Suppose that you start off with a file K of the keying signal and a stereo file B of the stereo signal to be processed. Use the mix option of Sound Designer II to mix two new stereo files BL and BR, where BL has a mono mix of K on the left channel and the left channel of B on the right channel of BL, and where BR has a mono mix of K on the left channel and the right channel of B on the right channel of BR.

In order to preserve stereo balance later, use 100% gains for transferring channels of B to BL and BR, DO NOT use scaling, and ensure that all time delays in the mixing are zero. Then process both BL and BR with the setup you have chosen, taking care not to change any control values when processing the two files (again to ensure stereo balance is preserved). The resulting processed files PL and PR may now again be mixed into a new stereo file P by putting the right channel of PL onto the left channel of P and the right channel of PR onto the right channel of P again using the Sound Designer II mix option. P will be the desired processed stereo file.

If one needs to mix the processed file with the unprocessed keying signal K, the last stage of preparing P may be bypassed, and the final mixed file M be obtained by mixing from PL, PR and K, putting the right channel of PL onto the left channel of M, the right channel of PR onto the right channel of M, and the original sound file K mixed and panned into M as desired. The C1 introduces no relative time delays between files even when lookahead is used, so that the files will stay in synchronism.

If the keying sound is on the right channel and the sound to be dynamically processed is on the left channel, click the Key Mode button to the $R \rightarrow L$ position. Where relevant, select the left channel signal before processing in this case.

Ducking

Ducking is an effect in which the gain of the keyed sound is reduced when the keying sound is louder than a user-determined threshold level. It is typically used to mix speech and background sounds where the background sound would normally drown out the speech, e.g. applause. If the speech or vocal sounds are used as the keying signal, and the applause signal gain is reduced in response to the speech, then the applause will be reduced in level enough that the speech will remain audible when the unprocessed speech is mixed with the ducked applause, because the applause level will be reduced during speech. Many similar applications exist where an important sound can be used to reduce the level of less important sounds to make it more audible.

For example, in recordings of amplified music, a mix from direct feeds from microphones and instruments can be used to duck ambience microphones, so that audience response is audible when the music is not playing, but the sound of the ambience microphone is reduced so as not to muddy the sound of the mix when the music is playing.

When using ducking processing, it is a good idea only to process the right channel (when using $L \rightarrow R$ key mode), so that the processed file has the unprocessed keying signal (e.g. speech or music) on its left channel and

the ducked signal (e.g. applause) on its right channel. The mix can then be done direct from the processed file.

The setups library provided two kinds of ducking. The first continues to duck levels further as the level of the keying sound (e.g. speech) increases further, whereas the second gives a fixed user-determined gain reduction when the keying sound is at any level above threshold. The second type of ducking may be preferred as giving a more natural result, but it is very much up to the user which is considered most appropriate.

Setup name: C1 Ducking 1

Here, the Comp/Exp module is used as a ducking device. The peaks of the sound on the left channel cause the right channel to reduce during those peaks.

The threshold level of the keying signal at which gain reduction starts is controlled by the Threshold control. The ratio control, which typically may be set between 2 and 50 or -50 and -15, controls the degree of ducking above threshold. The time duration for which the gain reduction lasts is controlled by the Comp/Exp Release control. The longer this is set, the fewer disturbing rapid level fluctuations will be heard in the ducked sound.

An added refinement is the use of the sidechain mode, so that it is the peaks in the equalized left channel sound that causes the right channel to duck. The above is actually tuned so that the right channel ducks during speech presence peaks on the left channel.

The active frequencies of the keying signal that operate the gain ducking may be selected by operating the filter type, frequency and Q controls. The effect of the equalisation on the keying signal may be auditioned by clicking the monitor button to Sidechain position. Keying by the wideband signal may also be selected by clicking the Comp/Exp EQ mode button to Wideband.

Setup name: C1 Ducking 2

Here, the Gate/Exp module is used as a ducking device. The peaks of the sound on the left channel cause the right channel to reduce during those peaks.

The threshold level of the keying signal at which gain reduction starts is controlled by the GateOpen control. The floor control, which may be set between 0 and +12 dB, (set to 12 dB in the above example) controls the ducking gain reduction above threshold. The output gain must be set at an equal and opposite compensating gain reduction (-12 dB in the above example) to ensure no gain when the keying signal is low in level. The output gain may, of course, be additionally varied about this value according to the user's requirements. The time duration for which the gain reduction lasts is controlled by the Gate/Exp Release control. The longer this is set, the fewer disturbing rapid level fluctuations will be heard in the ducked sound.

An added refinement is the use of the sidechain mode, so that it is the peaks in the equalized left channel sound that causes the right channel to duck. The above is actually tuned so that the right channel ducks during speech presence peaks on the left channel.

The active frequencies of the keying signal that operate the gain ducking may be selected by operating the filter type, frequency and Q controls. The effect of the equalisation on the keying signal may be auditioned by clicking the monitor button to Sidechain position. Keying by the wideband signal may also be selected by clicking the Gate/Exp EQ mode button to Wideband.

Ducked EQ

Ducked EQ is a variation of standard ducking, using the C1's bandsplit mode, where the gain reduction is confined to a limited frequency band. This is an extremely useful effect or tool whenever a mix suffers from the common problem of several sounds in a mix all occupying the same frequency range, so making sounds unintelligible. This setup creates "space" in the mix allowing a chosen predominant sound to become audible without losing the other sounds.

It may also be used in situations where conventional ducking is used (e.g. of applause when mixed with speech), where the ducking processing can be confined to a limited frequency band for reduced audible side effects.

Put the sound you wish to predominate in the mix on the left channel of a sound file and one or more of the clashing sounds on the right. Then using L -> R key mode and bandsplit mode for the ducking, ducking is confined to the active band of a filter tuned to the frequency band in which the aural clash occurs. Peaks in the intended predominant sound causes the level of the other sound to "duck" during peaks - but only in the problem frequency band. This minimizes the audible side effects of ducking, since the complementary passive frequency band remains unprocessed.

The interfering sounds duck in level only at the moments they need to. A typical example is a mix with saxes, electric guitar and vocals, all of which clash around 2 kHz. Putting the vocals on the left of the sound file and the sax or guitar on the right ducks the problem frequency range when needed to make the vocals clearly audible, while letting through these instruments the rest of the time and in frequency bands not needed for vocal audibility.

As in the straightforward ducking case, there are two kinds of ducked EQ setup provided in the C1 setups Library. The first continues to duck levels in the active frequency band further as the level of the keying sound (e.g. vocals) increases further, whereas the second gives a fixed user-determined gain reduction in the active frequency band when the keying sound is at any level above threshold. The second type of ducking may be preferred as giving a more natural result, but it is very much up to the user which is considered most appropriate.

Setup name: C1 Ducked EQ 1

Apart from the filter controls, this is used and adjusted in an identical manner to the C1 Ducking 1 setup described above.

The active frequency band in which the ducking operates is shown in red on the frequency response graph, and may be selected by operating the filter type, frequency and Q controls. The effect of the equalisation on the keying signal may be auditioned by clicking the monitor button to Sidechain position.

The ducked EQ may be bypassed by clicking on the Comp/Exp button immediately above the Comp/Exp bar meters.

Setup name: C1 Ducked EQ 2

Apart from the filter controls, this is used and adjusted in an identical manner to the C1 Ducking 2 setup described above, with one exception: The floor control, which may be set between 0 and +12 dB, (set to 12 dB in

the above example) controls the ducking gain reduction above threshold as before, but the makeup gain must be set at an equal and opposite compensating gain reduction (-12 dB in the above example) to ensure no gain in the active frequency band when the keying signal is low in level.

The active frequency band in which the ducking operates is shown in red on the frequency response graph, and may be selected by operating the filter type, frequency and Q controls. The effect of the equalisation on the keying signal may be auditioned by clicking the monitor button to Sidechain position.

The ducked EQ may be bypassed by clicking on both the Comp/Exp and Gate/Exp buttons immediately above the module bar meters.

Keyed Gating

Setup name: C1 Keyed Gate

Here the Gate/Exp module is used as a gate for the signal on the right channel, and the gating is keyed or triggered (i.e. opened) by the signal on the left channel. The gating is tuned by the sidechain filter and the GateOpen and GateClose threshold. The signal on the right channel is switched in time with the beat on the left channel. The longish release time shown here is to get a sense of "decay" after the beat, and the hold time is used to establish a minimum time for which the signal is switched fully on.

The GateOpen threshold may be moved up or down to adjust the triggering of the gate. Gate/Exp Release and Hold times can be varied to alter the "decay" of the percussion envelope effect.

The type, frequency and Q of the sidechain filter can be tuned so that the frequencies of the signal used for keying can be selected. The effect of the equalisation on the keying signal may be auditioned by clicking the monitor button to Sidechain position. Keying by the wideband signal may also be selected by clicking the Gate/ Exp EQ mode button to Wideband.

The keying can be bypassed by clicking on the Gate/Exp button immediately above the Gate/Exp bar meters.

Keyed expander

Setup name: C1 Keyed Expander

This setup is similar to a keyed gate, except that it replaces the gate by a low level (downward) expander. This has the effect of not cutting off the keyed sound during quiet passages, but of modulating its level with the envelope of the keying or trigger sound. This is particularly good for using keying to create a "new" percussion sound from a steady state sound like hiss or a synthesiser drone or even a natural sound effect like a babbling brook (be creative!). Unlike using a gate, a keyed expander has a more subtle and "organically natural" quality, due to its preservation of dynamic variations in the original keying signal.

Both the Gate/Exp and the Comp/Exp modules are set up as almost-identical keyed expanders in this setup, and either may be used for this effect on their own. But if both are switched in, the effect is more dramatic and effective.



The thresholds of either module (i.e. threshold and GateOpen) may be moved up or down together, by click-and dragging on the "button" between the Threshold and GateOpen Value Windows, to vary the threshold level below which expansion occurs. Release times can be jointly varied, by click-and-dragging on the "button" between the two Release Value Windows, to alter the "decay" of the percussion envelope effect.

The type, frequency and Q of the sidechain filter can be tuned so that the frequencies of the signal used for keying can be selected. The effect of the equalisation on the keying signal may be auditioned by clicking the monitor button to Sidechain position. Keying by the wideband signal may also be selected by clicking the Gate/ Exp EQ mode button to Wideband.

The keying of either or both of the two modules can be bypassed by clicking either or both of the bypass buttons immediately above the module bar meters.

Keyed EQ expansion

This is an especially flexible creative effect, in which a keying sound such as a percussion or drum track alters the equalization of a steady or continuous sound so as to impose the rhythmic percussion effect on the new sound. The flexibility lies is the ability both to tune the frequencies, and hence the tonal quality, of the sound that are effected and the exact degree, from almost subliminally audible to blatant, to which the EQ is varied in time with the original percussion. Additionally, the overall EQ can be varied in this setup.

This effect can be used to add varied rhythmic effects to continuous, drone or natural sounds that may not even be musical in nature, and can be useful in both musical applications and for musical dramatic effects in drama, documentary and other applications. In musical applications, the EQ can be tuned to ensure maximum audibility of the rhythmic effect of the keyed sound within a mix.

Setup name: C1 Keyed EQ Expander.

This setup is based on a keyed expander in bandsplit mode

The degree of boost of the selected active frequency band above the keying threshold is set by the makeup gain and the floor control, which may be set between +12 and -40 dB, must be set to an equal and opposite gain to ensure no gain in the active frequency band when the keying signal is low in level.

Any other degree of EQ in the active band above threshold can be used by setting makeup and output gains to equal and opposite settings. Additional overall adjustment of EQ in the active band at both high and low levels of the keying signal may be provided by varying the makeup gain above or below this setting.

The active frequency band in which the keyed EQ operates is shown in red on the frequency response graph, and may be selected by operating the filter type, frequency and Q controls. The effect of the equalisation on the keying signal may be auditioned by clicking the monitor button to Sidechain position.

The longish release time shown here is to get a sense of "decay" after the beat, and adds a kind of reverberant decay effect to the keyed sound.

The GateOpen threshold may be moved up or down to adjust the keying threshold. Gate/Exp Release time can be varied to alter the "decay" of the percussion envelope effect.

The keyed EQ expansion may be bypassed by clicking on both the Comp/Exp and Gate/Exp buttons immediately above the module bar meters.

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