Processors

Extra Phones in the control Room

The Beheringer headphone amp is labeled A and B on the patch bay. These inputs are normalled to Omni 1 & 2 on the board.
The input level knob is a master control for this input.

There are four output sections- note that there is a headpone out and Aux in jack on each section. It's easy to get them mixed up, but you will not hear anything from phones connected to Aux In. In addition to the front jacks, three of these sections feed remote jacks.
1 and 2 are in room 191 on the Mic panel.
3 is in room 190 by the door.

The Furman Headphone Mix System

The headphone mix system allows four individualized phone mixes in the studio. (There are 2 jacks per mix).
The head end is a box just below the patch bay. (Since the transformer in this box hums a bit, you will often find it turned off. the power switch is on the right back, open space is left in the rack so it can be reached.) There are 6 inputs. A stereo pair is connected to the console Studio Monitor output. Four additional inputs are normalled to the Omni outputs.

There are four headphone stations in the studio. These receive all six signals, and the mix is adjusted by controls on the box.

Using the Dominator Multiband Limiter

A peak limiter is not a compressor. A limiter is designed to keep signals from exceeding a particular level, either to prevent digital clipping or broadcast over modulation. (or cut through if you are still into making vinyl LPs).

A compressor is an overt processor used to noticeably change the sound, adding "punch" or the perception of loudness. The console has all of the compression we are likely to need, although taste is such an issue in compression, we may eventually add some outboard compressors to get a particular sound.

The Dominator is a limiter that operates independently in three frequency bands. This gives a smoother limiting when needed, and prevents high frequency peaks from "pumping" low frequency sounds and vice versa.
The Dominator is connected when needed at the patch bay. It has two main applications:

• When recording a classical ensemble with unpredictable levels. This means anything with percussion or possibly piano. Since the classical clientele will be primarily interested in the nuances of the performance, you don't want to be in the position of saying "Yes you played it better that time, but we can't use it because you clipped in your enthusiasm." In such situations, you will probably be using two mics patched to the dominator and then to the board. (Not possible until the patch bay is built, but the temporary setup will work nearly as well.)

• When "Mastering" a heavy pop mix that is destined for airplay. In that case, the dominator should process the output mix. Compression should be applied in the board track by track.

There is a good manual in the studio. Here is a quick synopsis of what the controls do:

- Input Gain- adjust this to set the level for the mid frequency band. To squash the sound, turn it way up. To just catch peaks, center or maybe lower.
- Process In/Out- use this to A/ B the effect so you know what is coming from the device, and what is already in the signal.
- LF EQ- this is a separate level trim for the low frequency section. This affects the bass when there is no limiting. If you turn it up, you will hear more bass, but the limiting will keep it from getting out of control.
- LF Crossover- defines the low in low frequency. 100 hz will catch only bass, 210 will include everything below middle C.
- HF EQ- trims the level of the high frequency band. Turning this up will give a brighter sound when there is no limiting.

- HF Crossover- sets the HF frequency. At 1.7 hz it will limit vocals somewhat independently of the bottom of the mix, at 3.4 hz vocals will be in the mid band and limited independently of the highs.
- Release time- sets how quickly the unit stops limiting. Start with the shortest time and adjust this for minimum gain pumping or loss of reverb. Note that the extreme setting is 7 seconds, very long.
- Stereo Coupling- keeps the image stable if limiting cuts in on one channel and not the other. Turn it off if you are only using one side.
- Density- There is overall limiting as well as the individual bands. Density adjusts threshold and attack time for this together. Higher settings give a louder effect.

Peak Ceiling controls-- the peak ceiling is the point where limiting occurs. It is adjusted by three controls:

1. Coarse-- steps from +2 to +24 in 2 dB steps. (that would be from -22 to 0 on the digital meters.)
2. Fine- fills in that 2 dB.
3. Range- 10 is designed for low level signals. This room is wired for +4, so leave the switch off.

The meter shows the limiting action, not the signal levels. The more segments are lit, the more the signal is squashed. Generally, if you are just using this for safety limiting, you only want to see occasional flickers.

PR from the makers:

Why is The Dominator So Special?

A very significant problem with wideband processing is "spectral gain intermodulation" which occurs when one part of the spectrum controls the level of another part. A typical situation is a vocalist being "sucked down" every time the kick drum hits.

Since most energy is contained in the lower frequencies, they tend to control the level of the entire spectrum. When the lower frequencies are above the limit threshold the higher frequencies are attenuated thus causing the output to be dull.

MultiBand processing solves these problems by splitting the audio into two or more frequency bands and processing each band separately. However, more bands often result in many more parameters to control including a method of summing the bands together again. While giving the user flexibility, it also requires different settings for almost every different source.

The Dominator II uses program dependent, intelligent circuits to reduce the number of controls. The user, therefore, has flexibility to shape the sound while quickly and easily achieving the goal of consistent, effective limiting.

The Secret Ingredient: ALT

A MultiBand processor splits the audio into separate bands, limits each band individually and then sums the bands together again. Even though each band's peak output is predictable, summing the bands together produces an unpredictable peak output.

One conventional approach to making the summed output predictable is to use a wideband limiter after the summing. This, however, introduces all the drawbacks of wideband limiting discussed above.

Another approach is to use a clipper on the summed output. This causes too much clipping distortion if the summed output is too high. In order to avoid this distortion the limiters' thresholds are set very far below the clipper threshold. The drawback is a loss of loudness and, due to the lower thresholds, much greater amount of processing.

The Dominator II uses a patented method to produce a predictable peak output while maintaining maximum loudness without audible distortion- the Automatic Limit Threshold (ALT). The outputs of the three bands are summed and sent to the ALT detector circuit. If the sum exceeds a reference value, the ALT reduces the thresholds of the individual limiters. When the summed output falls below the reference value the limit thresholds return to their original setting.

The ALT circuit has a self-adjusting finite attack time. The amount of time it takes to lower the thresholds of the limiters is the length of time the limiters' overshoot may be in the clipper. The reference value of the ALT in relation to the clipper determines the depth of clipping.

Both parameters are set by the Density control. When the Density control is set higher, the ALT reference gets closer to clipping, and the attack time is slower, producing more clipping. The opposite occurs when Density is set lower. The "0 RCH" position for the Density control emulates the standard parameters of the original Studio Dominator Model 700, and is recommended for general use.

It should be noted that there is only one ALT circuit controlling both channels equally. This provides global stereo balance and imaging by assuring that both channels always limit at the same threshold. This does cause an interaction if the Dominator II is used as two independent channels. Therefore, we do not recommend such a practice.

Notes on the Behringer Ultra Dyne Dynamics Processor

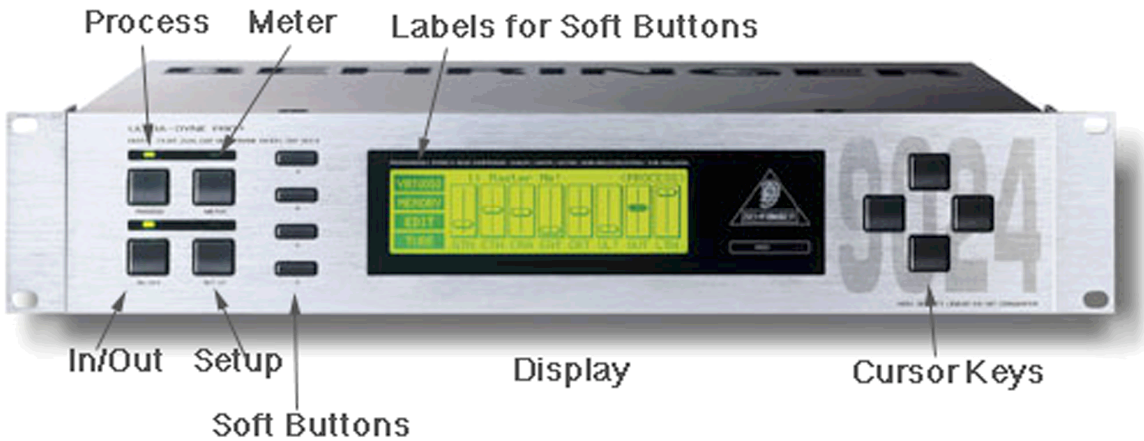
Note- if the unit is left on overnight, the display heats up and becomes unreadable. Turn the power off when you are done.

This is basically a multi (6) band compressor. It also includes a gate on the input, and a peak limiter, a tube emulator and an imitation Aphex exciter on the output. Since this is an all digital device, and is not necessarily real time, it can tell the future, which is handy for a compressor.

It has three levels of control:

- Factory Presets
- "Virtuoso" with automated program analysis and set up
- Manual tweaking

Here are the controls:



The In/Out button bypasses the unit.

The Process button gets you to preset selecting and adjustment. These adjustments are modifications of the settings of the current preset.

The meter button puts it into a display mode for actual use.

Setup is used to set the frequencies and compression of the individual bands and global processes like distortion. These settings are saved in the presets. Pressing the button chooses input/output settings, then steps through the bands. Holding the button gets to a deeper menu to change the input (once the digital option is installed.) and some things best left alone[1].

The meanings of the four soft buttons vary according to what page is in the display. The effect of each button is shown to its right.

The cursor keys navigate around the window. Sometimes you put the cursor on the name of the window to change pages. Sometimes the up and down change values of parameters.

Process

To call up a preset:

4. Press the memory button, then the load button. A window will appear with the name of a preset.

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[1] Does this thing really need a password?

5. Use the cursor up and down to scroll through presets. You can temporarily hear a preset by pressing the listen button.
6. When you find one you like, press the OK button.

The <MEMORY> page also lets you clear the current settings to factory defaults, store the current settings in a new preset, or compare the current settings with the presets.

General Tweaking

The controls that show in the<PROCESS> page apply to all six bands. To change a control, cursor left or right to choose it, then up and down to adjust it. The controls are enigmatically named:

GTH-- gate threshold -90 to -40 dB, off
CTH-- Compressor threshold -70 to 0 dB, off
CRA-- Compression Ratio 1:1 to 88: 1, infinite
CAT-- Compression attack time 0 to 250ms
CRT-- Compression release time 0.05 to 5 seconds
ULT-- Ultramizer (an automatic gain control) (amount) 0 to 100
OUT-- Output gain -24 to + 24 dB
LTH-- Limiter threshold (overall peak limiting) -36 to 0 dB, off.

Press the Edit button to set these independently for each band. (There is a parameter called band link in the settings- if this is on, you can't set the bands individually.

Press the TUBE button to add distortion. The sliders control:

EXC-- exciter process-- something like Mix on the Aphex Aural Exciter
O/E-- odd/even harmonic ratio-- something like Timbre
PRC-- Tube Process amount
TYP-- tube type- four kinds[2] of classic tubes are emulated.

The VIRTUOSO option will attempt to make threshold settings for you:
- Choose the amount of squashing you want from light to ultra
- Play your track
- Press OK

The machine measures the peak levels in each band, and sets compression threshold to give the effect requested. It might be a good way to study the different styles.

Meter Mode

You can check out the input or output levels, or watch the effects on all the bands.

[2] As if the type of tube was more important than the circuit design. But as a rule of thumb, the order in which the types are presented (12AX7, 12AY7, EL34, EL84) represents lower performance in fidelity, or I guess, increasing tubishness.

There's some good editing in meter mode- press the edit button or a cursor to start it:
- Buttons A or B will choose parameter
- Cursor left and right choose bands
- Cursor up and down adjust values.

You can solo band by band when you edit this way.

Setup

This menu gives more detailed control of the parameters, since you see the numbers. The Cursor keys take you around the window and the buttons give big step up, little step up, little step down, big step down, indicated by subtly different plus and minus signs.

You move through pages by hitting the SETUP button again. There's a page for input output settings, and for each band.

Input/Output

Some of these parameters are in the Process menu. New ones are:

LEFT or RIGHT If the unit is not in stereo mode you can set each channel. This picks which channel you are setting.

INGAIN Input level. I'd prefer a knob.

LIM. REL. Release time of the peak limiter.

Band Settings:

You choose the band by cursoring to the number and hitting soft keys.

LO FREQ
HI FREQ note that the adjacent bands will always have contiguous ranges, so if you edit the high frequency of band 1, you are changing the low frequency of band 2.

Noise gate settings
THRESHOLD--- signals softer than this will be cut out.
HOLD TIME-- the minimum time the gate will be on once it kicks in.
RELEASE TIME-- how the signal fades back in once the loud stuff returns.
PEAK WIDTH-- this instructs the noise gate to ignore short sounds like pops and clicks. It can do this because the gating algorithm is looking ahead across the delay time.

EXCITER DRIVE- bands 4, 5 and 6 drive the exciter. This sets the balance of effect between the three. It's a bit like tuning the Aphex.

Compressor settings are the same as in the Process screen, except for KNEE, which lets you adjust how quickly the compressor kicks in. Note that signals above the threshold are always fully compressed, but with a large knee setting, the effect will actually begin several dB softer.

Global Settings

You get to this menu by holding the setup button briefly.
VIEWING ANGLE: adjust for best legibility seated and standing.
INPUT-- Once we install the digital input option, this will choose between inputs.
MODE --- in stereo mode, all controls apply to left and right equally. That's usually a good idea with a device of this type.
DELAY-- enables the ability to predict the future. This delay should be set at 0 if you are processing signals live, or are processing one track of a mix.

SECURITY-- Leave this alone please.
PROTECT MEM-- this too

MIDI stuff-- Not connected.

Manual
The manual is a good read, as it not only has details about how this thing operates, it has some operating tips and a good bit of technical background

The Aphex Aural Exciter.

An exciter is a processor that adds "punch" to a sound by deliberately distorting it.. Think of it as a fuzz box for vocals. The Aphex[3] does a more subtle job than a guitar fuzzof course, but it is handy for helping a light voice cut through a thick mix.

Just exactly what this box does is a closely guarded[4] secret, but the main effect is the addition of high harmonics to a signal. That can sound nasty if it is overdone, so most of the features are designed to control details of the effect and prevent undesired side effects. As near as I can tell from the repair manual, this is what happens:

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[3] This is such a famous product from Aphex Systems Ltd that their name has become permanently associated with it.. When someone refers to an "Aphex", this is what they mean, even though they make many other devices.
[4] Well, you can figure it our from the schematics, but the usual description is buried in PR.

- The signal is filtered to isolate the band to process.
- The filtered signal is applied to the detector of a noise gate (but not processed yet.)
- Next comes a "transient processor" which seems to be a compressor with a fairly slow attack. This lets the initial peak through, then clamps down so the harmonics generated by the next section will happen mostly at the beginning of the sound.
- The dirty work is done by a "waveform generator" which is a pretty standard distortor[5]. This is blended with the compressed signal.
- This very rich mess is then used to amplitude modulate the filtered signal. Amplitude modulation produces sum and difference frequencies (sidebands) of the two signals. Modulating a signal by itself will produce sidebands that are all harmonics of the original. This modulation is applied through the noise gate, so nothing happens when the signal is quiet.
- This all gets mixed back to the original signal, hopefully in a tasteful amount.

To manage all this you have seven knobs and a bunch of buttons:

- **NR Threshold** adjusts the gating of the distortor. You should not hear any processing of background noise (including bleed from other instruments). If you do, raise this level. The red light should come on between vocal passages. When the singer is singing, the green light should come on. The Mode switch adds a second type of reduction[6] that may work better with hum type noise.
- **Tune** adjusts the low end of the frequency range that will be processed. Set this to be above the highest note the singer sings. (hint: 700 hz is F at the top of the treble clef.) You want to process the formants of the voice, not the fundamentals.
- **Peaking** adds a Q bump to the process filter. This will increase the amount of processing you get. The bump is about an octave wide at highest setting.
- **Null Fill** compensates for a side effect of the peaking control- in addition to a peak at the filter frequency, there is also a dip an octave lower, so if the singer sings high notes, the effect may pop in suddenly at some magic pitch. Null fill smoothes this out, so that as the singer moves up in pitch the effect comes in gradually.
- **Harmonics** sets the amount of extra harmonics generated.
- **Timbre** varies the mix of even and odd harmonics generated.
- **Mix** adds the harmonics back into the original signal.
- The AX switch turns the processing on.
- The Solo switch mutes the unprocessed signal, so you can hear exactly what you are adding.
- **SPR** stands for Spectral Phase Refractor, a device that compensates for the bass phase delay introduced by some analog audio equipment, such as tape recorders. Once of the reasons our all digital studio sounds so good is it lacks this problem, so SPR is no longer necessary.

[5] A half wave rectifier for you electronics buffs.
[6] Noise cancellation, like those fancy headphones for airplanes.

Using the Aphex

You don't want to use the Exciter on original tracks. Record as usual, and experiment with various settings during mix down.

7. Route the vocal track to buss one- do not assign to L-R.
8. Do not assign buss one to L-R either.
9. Patch buss one out to Aphex in.
10. Patch Aphex out to input 24.
11. Do not assign input 24 to L-R
12. Route input 24 to an unused RADARtrack, such as track 24.
13. Assign input 48 (where track 24 comes back to the board) to L-R.

If you put RADARtrack 24 into record ready (auto record off) you can hear the effects of the Aphex.. You may want to record several tracks with different settings and see which works best in the final mix.

Usually all you need is a bit of edge to the voice. This will be noticeable when the track is heard alone, but will disappear when the mix is brought up. If everything works right, the voice will still sound natural, but you will hear it better.

Here's some PR from the Aphex web site:

> The Aphex Aural Exciter Type III utilizes a patented audio process that will recreate and restore missing harmonics. These harmonics are musically and dynamically related to the original sound. When added, they restore natural brightness, clarity and presence, and can actually extend audio bandwidth. These harmonics are so low in level however, they add little power to the signal. Unlike an equalizer or other "brightness enhancers" which can only boost high frequencies, the Aural Exciter Type III extends the high frequencies. It is a single-ended process that can be applied at any point in an audio chain, and needs no decoding.
>
> The Aural Exciter process consists of two audio paths. The main path and the process sidechain path. The main path transparently conveys the audio signal directly from the input stage to the output stage, maintaining unity gain with wide dynamic range. The sidechain path contains all of the Aural Exciter processing circuits and receives audio from the input stage.
>
> A mixing circuit in the main path allows the sidechain output signal to be mixed with the main signal. The user adjusts the amount of "MIX" to set the strength of the effect.
>
> Two Modes of Noise Reduction are provided with the Aural Exciter Type III, allowing it to provide enhancement without adding to the noise floor of reasonably noise free sources. It can also enhance the brightness,

detail and clarity of seriously noisy audio sources while, at the same time, erasing much of the original noise.

Mode "A" operates as a linear sidechain expander with variable threshold. The expansion ratio of 2.5-to-1 permits the sidechain "MIX" to follow the signal level below threshold, so when the higher frequencies of the input drop below threshold, the "MIX" will drop at a proportional 2.5 to-1 ratio. Thus the original signal-to-noise is not affected even with a great deal of enhancement. Mode "B" is a revolutionary new noise reduction technique which operates to actually "erase" source audio noise while the Type III enhances the signal. This allows you to restore brightness and intelligibility to noisy recorded tracks or other noisy sources and improve the signal-to-noise ratio!

Both modes are fast and easy to use, and effective in any application from live sound to broadcasting and recording.

The "SPR " (Spectral Phase Refractor) function of the Type III is a totally new concept in psychoacoustic enhancement which can produce some amazing results. Through the many steps of recording, duplicating, distributing and reproducing sound, the phase of the low frequency audio spectrum becomes delayed compared to mid and high frequencies. This is a natural and unavoidable effect which becomes worse with each generation.

When the bass frequencies become delayed in time compared to other sounds, the bass loses fullness and definition and seems to become less powerful even though there may be no actual loss of bass frequency response. The high end also loses definition, seeming to get duller.

Amplitude equalization at this point will not fully restore the clarity and bass power. indeed, it may worsen the condition causing clipping or overload distortion. The "SPR" corrects the bass delay anomaly to restore clarity and openness and significantly increases the apparent bass energy level without adding any amplitude equalization or "bass boost".The "SPR" function works harmoniously with the new Aural Exciter circuitry to give the Aural Exciter Type III amazing new capabilities.

The "Drive" control of previous Aural Exciters has been eliminated and Adjustable Harmonics Mixing of the exact harmonics level desired is now available making the Type III more flexible and easier to use.

Null Fill is a new and useful tuning adjustment introduced for the first time with the Aural Exciter Type III. The addition of NULL FILL to the PEAKING and TUNE controls gives the Type III more power and flexibility to enhance all types of audio sources. To understand how it works, it is necessary to understand a physical phenomenon called "Phase Nulling" which occurs with all Aural Exciters when the MIX control is adjusted to obtain high frequency enhancement.

There is a time delay associated with the sidechain signal which is an important part of the operating theory of the Aural Exciter. This time delay causes transient waveforms to be slightly "stretched" as the sidechain signal is added to the main audio path. The "stretched" transients are then perceived by the ear as more pronounced or "louder." The side effect of the time delay is a "dip" or "null" in the output equalization curve. The null can be a desirable characteristic because it compensates for the slight additional power added to the signal by the high frequency shelving boost. The null de-emphasizes the frequency range around the TUNE control setting, thus giving even greater emphasis to the higher frequencies.

There are times, however, when the Phase Null is unwanted. The NULL FILL control allows the user to "fill-in" the phase null to any desired amount, thus further improving presence.

Servo-Balanced Inputs and Outputs Although more costly than conventional designs, servo-balancing offers many advantages. Servo-balanced input circuits absorb high common-mode voltages found in long cable runs without sacrificing headroom. Servo-balanced outputs are not only short-circuit proof, but can be used single-ended at any time without the usual 6 dB loss of conventional circuits. To use input or output single-ended (unbalanced) the user need only ground the unused pin. Inputs and outputs are fully RF protected. The unity gain 1/0 structure is normalized for both + 4dBu (professional) and -10dBV (IHF) operating levels by switch selection from the rear panel.

Typical Applications

The Aural Exciter may be used in many ways for audio enhancement. Depending upon the requirements, either pre-or post-processing may be selected. Either source optimization or system optimization or a combination of both is possible. For example, a PA. system may be greatly enhanced by using the Aural Exciter to increase the intelligibility of the loudspeakers, thus improving penetration of the sound around corners and in areas usually difficult to fill. In another case, the source may sound dull and hard to understand. The Aural Exciter will compensate for this deficiency by adding brightness and clarity to the sound more effectively than use of equalization alone.

The Aural Exciter may be used to "pre-process" recordings to anticipate the audio degradation in the medium or during subsequent reproduction. Much of the detail added by the Aural Exciter will survive filtering and distortion of the reproduction equipment, and provide a better quality audio playback. Audio and Video Cassette duplication are examples of this application. Broadcasting is another good example.

In the recording studio, post production suite or similar environment, post-processing of previously recorded sound tracks can restore lost vibrance and realism, even to the extent of saving dialog or sound effects which were thought to be unusable. Instruments and vocals can be made to stand out in the mix without substantially increasing the mix levels or using equalization.

Many electronic instruments are limited by their sampling rate (bandwidth) and word length (resolution), they can sound lifeless. The Aural Exciter actually extends bandwidth and adds details making synths, samplers and drum machines come alive.

Video and film audio are both bandwidth limited and compressed. The Type III is especially useful in creating the perception of higher frequencies and greater dynamics with pre processing, thus bringing more presence and clarity to the final product.