K-Stereo™

Operator's Manual

MODEL DD-2

Manufactured and Marketed by Digital Domain, Inc.

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Patent has been applied for. Processing is Patent Pending

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A LETTER FROM THE INVENTOR

Welcome to a brand-new category of audio product: **the world's first *Ambience Recovery Processor*.** You are about to experience a sound enhancement process that has been in secret development for over 10 years, exclusively in use at *Digital Domain's* Mastering Studio in New York and Orlando. K-STEREO™ (patent pending) is a unique, mature product that has been in continuous refinement. The latest "deep" algorithms, which I developed in the year 2000, are 3 to 6 dB more effective than the ones I produced in 1990!

This box is the ultimate sound *polisher.* For the first time, the mastering engineer can enhance, expand, and equalize the ambience in a recording. The nicest thing about K-Stereo is it works **unobtrusively and naturally.** Do you have a "big" rock and roll recording that's too "small"? You can fix it, avoiding a costly remix. You will find K-Stereo an *extremely natural-sounding* stereophonic and surround enhancement technique during post-production, mastering or even mixing. K-Stereo gives you control over your reverb returns or ambience mikes *after the program has been mixed*, and it enhances the shape, spread, and depth of that reverb. Reverberation envelops the direct sound but the sound actually becomes clearer. If you think that is a contradiction in terms, then you're in for a surprise.

When I first started using K-Stereo, I only used it on masters which were obviously lacking in space or depth. But now I try it on most every master that comes in, because of the clarifying effect of the process---instruments become a little clearer and you can hear more detail in the mix, without having to add any exciters or change the tonal balance. Thus, currently about 80% of the masters that leave Digital Domain's studio have been K-Stereoized to some extent.

Try it yourself. Recently, we had a *K-Stereo Party* and tried it on dozens of older pop-music CDs. We all agreed it enhanced at least 5 out of 10, some with spectacular results!

K-Stereo is unlike any of the so-called "reimaging" processors you have heard. While other boxes seem to have a sound of their own...this processor fills in what's been missing. K-Stereo initially appears subtle. Even when you turn it all the way up, it doesn't sound bad. Don't expect an *out of body* experience! Instead---you'll hear a natural, diffuse ambient field derived from the ambience in the source recording. K-Stereo enhances a recording's clarity, spatiality, depth and soundstage without creating phasing or comb-filtering effects, without matrixing or altering mid/side ratio, without changing the "mix", without fancy steering, and with no effect on the tonal balance of the direct sound.

Wordlength reduction: Normally, a 16-bit reduction tends to have less ambience and spatiality than the original. But through the use of the K-Stereo process and POW-R™ dither, amazingly, a 16-bit master can sound better than the 24-bit originals! And a 24-bit master will sound even better

than that! I state all this with confidence, after 10 years of success and many happy clients. Enjoy!

Bob Katz

P.S. **This is an all-digital processor.** If you are working with an analog audio system you will first need to connect one or more A/D/A converters to this processor. Try to use converters with the longest usable wordlength (20-24 bits) and highest sample rate (up to 96 kHz) for the best sound.

P.P.S. **If you're already familiar with the operation of the DD-2, note that the latest software, November 2001, includes an ergonomic enhancement which blinks indicator lights if any control has been altered from its default. For example, if the K-Effect is turned on, the LED next to K-Level will blink when that button is not pressed.**

QUICK SETUP

K-Stereo lets you enhance the level, depth and soundstage of the original ambience in previously mixed material. We're sure you want to hear this new process, *so let's get started...*

A. Stereo enhancement of musical program: It's real easy to use. Just connect this digital processor in series with any stereo linear-PCM AES/EBU audio signal chain, turn it on, and load factory preset #00. Load preset 00 by pressing the **PRESETS** button, rotate the **left knob** to position 00, and turn the **right knob** until the letter "A" appears after the word "LOAD". The screen should now look like this:

```
A=00<   B=(anything)  LOAD A
```

Press the **K-LEVEL** button. Turn the K-Process ON by rotating the right hand knob. NOW, LISTEN! *While listening*, adjust the K-Level with the left knob. Make sure the right hand side of the display says "ON", or rotate the right knob until it does. K-Levels from -2 to +1 dB are typical settings, but you may exaggerate the level up to +6 for educational purposes or for special effects. As you become a *stereoization expert* you will learn that K-Levels as low as -3 to -5 can produce a subtle but meaningful enhancement on the most intimate material, especially acoustic solos, duets, trios, etc. Press the **BYPASS** button to bring on "withdrawal symptoms" and educate your ear to the power of this subtle, but important enhancement process.

For those of you who've been skeptics about the virtues of long wordlength recording over 16-bit, the K-Stereo processor will educate your ears to the improved ambient quality that can be encoded by a native 24-bit recording. But with K-Stereo, you can bring out ambient detail in any wordlength recording.

Since true *ambience recovery* is program-dependent, audition many types of music selections to hear the range of the effect, which can vary from subtle to dramatic, depending on the inherent reverberation character of the recording. K-Stereo is often *over-dramatic* on spoken word, often revealing the defects in typical announce booths, so little or no enhancement is possible with spoken word, perhaps a K-level of below -6 or off. It helps to begin with unprocessed master tapes, since many contemporary pressed CDs are (sadly) hypercompressed (they sound like they're going through a bad FM radio to begin with!), and K-Stereo doesn't necessarily help a distorted "mishmash".

B. Surround enhancement: Model DD-2 can be used exactly as above to enhance the front channels of a surround master.

That's it in a nutshell. Now let's investigate the subtleties and possibilities of this fascinating processor.

K-Stereo Features

Stereoization: K-Stereo allows you to *reshape an original mixed recording* in ways not previously possible----without directly affecting the original vocal/instrumental mix. Instantly create a deeper, wider soundstage (adjustable in four different modes, from subtle to dramatic effect), move instruments and vocalists backwards (away from you and towards the front wall). Instruments and vocalists take on a spatial, kinetic, quality, as if they had been naturally recorded with stereo microphones in a real space. There is no ugly coloration, pitch modulation, “phasiness” or artifacts often noted with other processes. You can restore or enhance the missing front-to-back depth in stereo recordings that otherwise sound flat and unidimensional. Recordings become *integrated* and *organic;* “in your face” recordings can be made to sound just right. Adjust the degree of the effect according to the mood of each music track; it can range from *subtle and intimate* to *extremely spacious* with the touch of a knob, without losing center-panned information!

K-Stereo also performs the most natural mono-to-stereo conversion, by bringing out the ambience in the original mono source and spreading it to the sides in a stereophonic fashion. And the result is extremely mono-compatible.

MIDI-based snapshot automation allows you to remotely control the process with precise timing (external MIDI sequencer required), and store/recall your presets (SYSEX read/dump function).

Ambience Recovery and Enhancement: K-Stereo *extracts* the original ambience in recordings, enhances and spreads that uncorrelated ambience between and beyond the loudspeakers. The result is a large soundstage that is enveloping and unfatiguing. It’s exactly like having a handle on the reverb returns in an original mix *after it’s been made*. Dry instruments remain relatively dry and wet instruments get wetter as you turn up the **K-LEVEL** (ambience control). You will be able to hear details in the reverberation that were originally masked, increasing the illusion of a real space. This is accomplished **without adding artificial reverb**, without the muddy effects and artifacts that are associated with adding reverberation to an existing mix. Plus you won’t have to spend an hour playing with reverb settings to avoid the “room within a room” effect, or attempting to match the color of the original recorded reverb.

Do you long for that *big* sound? Poor monitoring environments, especially nearfield monitoring, can result in recordings which sound too dry and small. Recordings which have been truncated by going through too many generations of 16-bit media tend to lose their spaciousness. Cheap reverb units produce little dimension or space. During mastering, you can enhance recordings made by less experienced mixing engineers, or even mixes made by expert mixing engineers!

K-Stereo improves the spread and diffusion of the inherent reverb. It can’t turn a

$600 reverb into a $6000 model, but it can give the $600 reverb some of the qualities of the more expensive model.

K-Stereo can help give a demo the polish of a better mix, or reduce the weaknesses of a poor mix. There's no substitute for a great mix; no substitute for use of stereo and surround microphone techniques in a natural acoustic space----- however, an experienced mastering engineer will discover that K-Stereo can put the final polish on nearly *any* mix.

Note: The psychoacoustic nature of our process requires the direct sound source to function, so you will not hear the effect on a "pure reverb return".

Increased master clarity: Dense stereo recordings often sound "congested". Some of this is due to resolution limits of the digital medium, and to the limited physical space between the two front speakers. K-Stereo helps reduce that congestion, revealing inner details that were previously masked. Each instrument and vocalist becomes better defined in its space. Instead of getting harsh, the sound gets both **warmer and clearer**, which used to be a contradiction in terms! Recordings that were made with multiple stereo microphone techniques sound more realistic and cohesive, since K-Stereo can remove the artificial "edges" between the different spaces you're trying to mix together.

USES FOR K-Stereo

Music Recording: K-Stereo is *ecumenical.* It enhances country music, classical, R&B, rock, acoustic and electric music. *Digital Domain's* clients have been "hooked" by this unique stereoization process for over 10 years.

Surround Sound Producers---Eliminate 2 Mikes and 2 channels from your recording endeavor! Classical recording engineers are currently using various methods of recording surround via 4, 5, 6 or more original channels on location. But it's important to consider stereo compatibility, the countless audiophiles and critical listeners who currently have (only) 2 channel stereo systems. Often the channels, extra mikes or floor space are not available to place a special additional front pair "just for stereo".

The front microphone pair from your surround recording will likely be a bit too focused, intimate, sharp or clear than you would want for stereo recording. But with K-Stereo, there is **NO COMPROMISE**. The Model DD-2's ambience extraction ability will, extract and reshape the inherent ambience in the front mike pair, and produce a stereo recording with depth, space, definition, and clarity---a recording that will be EXACTLY as the producer would desire in a stereo recording.

Film Scoring: We have successfully used K-Stereo to enhance an orchestral film score that was recorded in a pleasant acoustical environment, but it was not wet enough, plus, the multimiking technique did not have the spatial qualities possible with minimalist miking techniques. In this case, we put the model DD-2 in front of an artificial reverberator. The two worked hand in hand to produce a natural orchestral sound, with K-Stereo enhancing the stereo image and as much of the original reverberation as possible. Then we needed only a touch of the artificial reverberation, avoiding the “room within a room” effect. K-Stereo can also be used at the final film mix stage to increase the depth, breadth and spaciousness of the orchestra when the score or scene requires it, or make it more intimate to avoid competition with the dialog. All these changes can be automated with a MIDI sequencer against timecode.

“Analog” processing simulation: K-Stereo can replace or supplement analog (tube) processors which have often been used to warm or enrich sound quality or “loosen” the stereo image. K-Stereo is an *organic* process that successfully enhances sound without adding harshness or making “cold” sound---*we guarantee it.*

“Internet” Pre-Processor: K-Stereo can make a *small* sound *big.* Lossy coding processes (e.g., MP3, MPEG-4, Atrac, AAC, AC3), leech ambience, space and low-level information from recordings, make them sound *small*. Many engineers try to use compressors to reduce these losses, which “thicken” the sound and increase reverberation, but the “cure” is often worse than the disease. Also, lossy coders were not designed to deal with hypercompressed recordings, resulting in serious distortion and less satisfactory sound out of the decoder. K-Stereo can effectively deal with ambience and spatial losses without forcing you to resort to excessive compression of dynamic range.

K-Stereo will help you get that *big* sound through all the degrading radio processing and lossy Codecs. In the old days, we had to learn to precompensate for the losses of vinyl. K-Stereo is a pre-processor for the low-bit-rate digital age.

Video and Computer games are often played on small, compact systems. K-Stereo can help make recordings sound *naturally* bigger and more spacious on such systems without the ugly phasing and “out of body” artifacts that you’ve heard from other enhancement processes. You can also use it as a special effect for the music or effects in the video game, increasing the stereoization and widening the sound image at programmed moments in the material.

Stereo and Surround Synthesis: K-Stereo takes the original ambience in a mono recording and spreads it in a non-correlated manner to the left and right speakers. This produces a convincing and natural stereo effect that remains remarkably mono-compatible.

Restoration/Restoring Losses: K-Stereo is a genuine digital restoration process that can extract inner details from older recordings.

> **Digital Sources:** K-Stereo can *rejuvenate* 1630 masters or other 16-bit masters, from rock to classical, restoring much of the ambience lost through the inferior A/D converters of the time, or through multiple digital copies that have been truncated or improperly dithered. In addition, harsh-sounding multimiked orchestral recordings with a flat soundstage may obtain some of

that “Decca-Tree” quality with a touch of K-Stereo. There’s no substitute for retaining those wordlengths (bits), but K-Stereo restores much of the ambient information lost due to truncation. *We guarantee it.*

Analog Sources: When mastering from older analog tapes, noise reduction is often used to reduce tape hiss levels. No matter how sophisticated the noise-reduction process, some degree of ambience will be lost. K-Stereo digs deep into the original ambience of an analog recording and restores it to a remarkable degree.

Forensics: K-Stereo enhances the intelligibility of monophonically recorded spoken-word recordings made under adverse conditions in reverberant spaces. Because the original reverberation is now spread stereophonically, the ear is able to focus binaurally on the center-placed direct sound. You must listen in stereo for better results. Headphone-listening may be better for speech intelligibility under these circumstances.

An Enjoyment Box: The ear likes the tickle of a spacious sound quality. Use K-Stereo to improve the beauty and musicality of a recording. Not every recording sounds right “bigger”, so just like every process, evaluate your choice on a case by case basis. Too much of a good thing, you know! In mastering, we often turn off the stereoization for some album tunes and back on for others.

GUIDED TOUR---OPERATING THE UNIT

System Menu
Dither

All effect mixing must be performed within the DD-2. Do not externally mix the dry source with the effect from this box. Due to built-in processor delay, external "dry/wet" mixing would result in an ugly, comb-filtering effect. Due to the psychoacoustic nature of the ambience extraction process, we cannot provide a "wet only" signal for external mixing.

Model DD-2 is very easy to use. The best way to discover how this box works is to explore the menus and buttons yourself. Then, return here:

SYSTEM MENU

The System Menu sets the default parameters of the unit. Let's cycle through the system menu "screens". Each press of the System button cycles through the following screens:

DITHER

Rotate the left knob to cycle through several output wordlength and dithering choices:

`24 bits, 24 dithr, 20 bits, 20 dithr, 16 PWR-1, 16 PWR-2, 16 PWR-3, 16 dithr`

Normally you would use the dither choices at all times. The most common setting for mastering will be **24 dithered**, if you are feeding further digital processors down the line. Use the "24 bits" position ONLY if you wish to automate turning the dither on and off while mastering, for example, if you have a song which was already dithered and processed, then use the MIDI control to select 24 bits. In that case, make sure the K-Level is OFF, and all equalization is off (at 0 dB), and the box will then produce a perfect clone of the source, the same as BYPASS. If you have any process engaged and the output is set to 24 bits, the box will introduce quantization distortion. In other words: Use the "24 bits" setting with caution. When in doubt, leave the dither ON.

Always choose the longest output wordlength that the next piece of gear in line can accept. Use **20 dithered** only if recording to a 20-bit medium such as an ADAT M-20, and only if there are no additional processors between our processor and the recorder. We recommend using the 16 bit dithers if recording to a 16-bit medium such as a DAT, ADAT, DA-88, or the AKAI dubber, and only if there are no additional processors between our processor and the recorder. POW-R type 3 dither, licensed from the POW-R consortium, is highly acclaimed as the most transparent dithering process. Try it yourself; switch between POW-R and 24-dithered and see if you can hear the difference. It will be extremely difficult. We also provide POW-R 1 and POW-R 2 dithers, as well as a flat dither (labelled **16 Dither**). POW-R 1 is a Nyquist-band dither, and POW-R 2 uses a slightly less aggressive noise-shaping than #3. The audible differences are very small, with POW-R 3 providing the most resolution, and the lower dithers using less aggressive shaping.

Since the K-Stereo-processor will usually be used in front of other processing, you will probably not be able to take advantage of its 16-bit POW-R dither in one pass through your signal chain (remember: always feed the next device the longest wordlength it will accept). You could apply 16-bit POW-R dither on a second pass, turning off all the K-processing and EQ.

The 16-bit dithers are not available at sample rates above 48 kHz.

(24 dithered is the setting loaded into Factory Preset 00)

MIDI

The screen looks like this when you cycle to the MIDI screen (3rd system menu):

```
CH 1      READ     NO
```

The Left Knob changes the Midi Channel for Midi Program changes, Sysex READs or Dumps. The middle knob chooses either Sysex READ or DUMP. The Right knob, choosing either YES or NO, starts the READ or DUMP.

There is a Midi Receive indicator in this screen, an asterisk that blinks next to the channel indication. Use this for debugging.

For a sysex dump, first choose DUMP with the middle knob. Then start your sequencer or librarian recording on the proper channel, then rotate the right knob from NO to YES. The YES indicator will blink, and you should see some indication on your sequencer that MIDI information is being received. **For a sysex read** (restore of all Presets), turn the middle knob to READ, turn the right knob to YES, then play back a sysex file into our processor. When the processor receives the Sysex information, the display will count down the memories as they are restored. To cancel a READ before sending information to the box, turn the third knob back to NO. The READ cannot be manually cancelled once it has started.

PRESETS

LOAD PRESETS

Press the PRESETS button once to see the LOAD screen:

```
A=01     B=05     LOAD
```

Loading presets is very simple. Switch between any two presets in this screen. The **left knob** chooses which will be Preset A, **the middle knob**, Preset B.

Rotate **the right knob** clockwise, and the screen will change to this:

```
A=01<    B=05     LOAD A
```

This means PRESET A has been loaded, and thus memory 01 (in this example indicated by the caret next to preset A) has been loaded. Rotate the right knob in either direction to display LOAD B, and you will see:

```
A=01     B=05<    LOAD B
```

Indicating that memory 05 has been loaded. The right knob is a continuous controller, cycling between A or B as you rotate in either direction.

Normally the system is designed to alternate between A or B presets, so you change the inactive one to the memory you desire, and then LOAD it. However, if you change the number for a *currently active preset,* the caret will disappear, and the right indicator reverts to LOAD (as if you had just entered this screen). You must then rotate the right knob to actually load this new memory. A new memory has been loaded as soon as a caret (>) is displayed and the letter A or B is displayed on the far right of the screen. The Preset load dial is continuous, so you can get to memory 99 fast by rotating counterclockwise “below” memory 00. In addition, the Presets menu is context sensitive, remembering whether you last rotated memory A or Memory B when you do a LOAD or whether you were last in SAVE mode.

ALL FUNCTIONS OF THE ENTIRE BOX are remembered in the presets; INCLUDING THE SYSTEM MENU---EXCLUDING the MIDI channel, EXCLUDING the Master BYPASS. Be careful of this when reusing an old preset. After doing a job that uses a lot of registers, it’s a good thing to “clear the registers” by loading PRESET 00 into the series you’re going to use to return to a “Factory default”, or use one of your favorite defaults. You might store your favorite default into a register you are unlikely to use for other purposes, such as register 99.

FACTORY DEFAULT= PRESET 00: 24 dithered, Wide & Deep algorithm, all EQs flat, K-level at 0 dB, K effect is OFF. All other levels set to 0 dB.

When the box receives a MIDI program change command, this screen will display the new memory that it has loaded in both A and B registers, like this:

```
A=07    B=07    LOAD
```

SAVE PRESETS

Press the PRESETS button a second time for the SAVE screen:

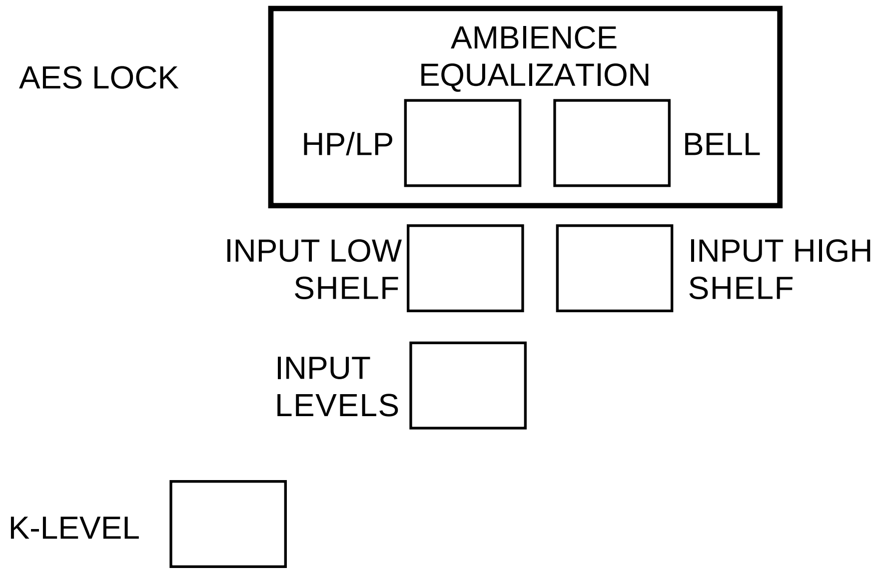
```
01      SAVE
```

To save a preset, rotate the first knob to the register number you wish to save the current state. Then rotate the second knob (in either direction) to save in this register. When the information has been saved in this register, the screen will blink, and the display will say

```
01      SAVED
```

You can easily save the current box state to a series of sequential registers for a starting default. Turn the first knob, turn the second, blink, repeat. (After you save a register, as soon as you move the first knob to any new register, the letter “D” in SAVED disappears). The last Midi channel used and the current state of the box are always saved in non-volatile, flash memory in case of power failure. Note: You cannot save anything into PRESET 00, which is a factory preset. Memories 1-99 are available for user storage.

BASIC USE CONTROLS



K-LEVEL

This screen is the heart of the unit. Now you, the mastering engineer can adjust the ambience in a recording.

Pressing K-LEVEL displays:

```
K 0.0    WIDE +  DEEP  ON
```

These tell you, from left to right:

- the K-Level (extracted ambience)
- the ambience extraction algorithm and
- whether the K-Effect is ON or BYPASSED

Adjust the K-Level (left) knob, in 0.5 dB increments, to affect the degree of ambience extraction and stereoization. 0 dB is a good level to start with. Delicate or intimate musical material requires less K-LEVEL, poorly recorded material, or special effects require more.

What does a K-level of “0 dB” mean? This is strictly a calibrated start point. I decided to call this level “0 dB” after years of experience that this is the most common K-Level to start at. There is no absolutely no problem using a “minus” or a “plus” setting as well. **A setting of K = -1 dB simply means 1 dB less extracted ambience than the reference of 0.** Of course, it becomes more and more subtle the more you turn it down. Most people give up after –3 dB and it’s real hard to hear an enhancement below about –3 dB with most material.

K-LEVEL
Algorithm
Ambience EQ

The limit of K-Level is + 6 dB. It is **extremely rare** to desire a K-Level setting greater than + 2 dB, especially with the *deep* algorithms. As a rule of thumb, you will need 1 or 2 dB less K-LEVEL when switching to a *deep* algorithm. Nevertheless, don't be afraid to use +4 to +6 dB of K-LEVEL if the material calls for it; just watch out for that "too big" sound. Instruments will take on a gorgeous sense of depth, stereophonic life and definition. Here's a secret: K-Stereo makes the sound louder, but rarely causes clipping. (shhhh).

Algorithm

Rotate the Algorithm (middle) knob to choose among four K-Stereo algorithms. These algorithms are:

```
SMALL
SMALL +  DEEP
WIDE
WIDE  +  DEEP
```

The WIDE + DEEP algorithm is the most powerful, with the best ambience extraction, and the widest and deepest soundstage. This is the algorithm permanently saved in factory PRESET 00. If you want less of an effect, try lowering the K-Level, and/or changing the algorithm. If a recording is already quite spacious width-wise, or if you do not want to make it more spacious width-wise, then try the SMALL + DEEP setting to increase the depth from front to back.

Rotate the right knob to turn the K-effect on or off.

Tips on Adjusting the K-Level: Adjust the K-level to your esthetics. Take advantage of the right-hand ON/OFF knob to test if you have "spaced out" too much. In most cases, you will discover you want a little less K-Level than you had during that initial flush of excitement! Remember to keep a proper balance amongst clarity, intimacy, and spaciousness (which equals "power and depth"). K-LEVELs below about -5 dB are extremely subtle and will rarely be used. -10 is effectively OFF to all but the most sensitive ear.

AMBIENCE EQUALIZATION

Now the mastering engineer can equalize the ambience in a recording. The most commonly used ambience EQ you will use is the High-Pass filter, which is a first order "bass cut" or "high pass" filter *on the extracted ambience only*. With this control, it's possible to tighten the spatiality of the bass instrument(s) without affecting the ambience in the midrange and highs.

We've provided two other ambience filters, a Low-Pass and a bell. These will be rarely used. We've occasionally used the Low-pass filter when material comes in that sounds harsh and bright and we want to prevent this distortion from "leaking" into the ambience. Since our ambience extraction algorithm depends on wide bandwidth, there is a point of diminishing returns. We've also provided a bell (parametric) filter on the ambience, which you may use occasionally, for example, to warm up the ambience. This can be powerful when working with poorly-recorded stereo masters. For example, we've successfully used a 225 Hz ambience filter with 1 dB boost/Q 0.7, in a situation where a vocalist sounded just fine and didn't need any warming up, but the instruments were a bit bright and harsh. Standard Equalization techniques don't work in this situation. But with K-Stereo, *you can selectively equalize the ambience or reverberation in a recording*. You may not be able to turn a sow's ear into a silk purse every time, but this tool can turn quite a few tricks.

Ambience HP/LP Filters

To engage these filters, press the HP/LP button, showing:

```
HP 100   LP 22K
```

The left knob adjusts the ambience High pass filter in ISO steps. The middle knob adjusts the Low Pass filter, and as we suggest, most times leave it OFF. The OFF positions for each filter are below the lowest frequency or above the highest frequency.

Ambience Bell

To engage this filter, press the BELL button, showing:

```
00.0    63 Hz   Q 0.6
```

The left knob adjusts the boost or cut of the bell curve, the middle knob the frequency, and the right hand knob the Q. In most cases, leave this level at 00.0, which is truly disengaged from the DSP (quality filtering courtesy of Dr. Glenn Zelniker!).

INPUT EQUALIZATION

Input Low Shelf

This is a convenient tone control for the input signal, using the low-distortion equalization algorithms developed by Dr. Z. It's a first order low shelf, boost (or cut). Press this button to see:

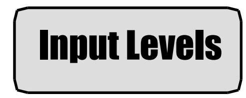
```
00.0   100 Hz   lo shelf
```

The left knob adjusts the boost or cut of the shelf EQ, and the middle the transition frequency, *below which action occurs*. This input EQ precedes all other circuits in the Model DD-2.

Input High Shelf

In some cases you may desire a touch of high frequency boost on the input source when you add restored ambience, or conversely, you may wish to “soften” an overbright source. We provide a first order shelving boost (or cut) for this purpose, and to make it easy to compare the total effect of EQ and ambience with a single BYPASS button. If you need more control than the two shelves, use an external equalizer. This input EQ precedes all other circuits in the Model DD-2. Press this button to see:

```
00.0   2K25 Hz  hi shelf
```



The first knob adjusts the boost or cut of the shelf EQ, and the second the transition frequency *above which action occurs.* With these two shelving filters, K-Stereo and dither, you may well find this box has everything you need to process a well-recorded master!

INPUT LEVELS

This button cycles between two screens. The first screen...

```
M 00.0   S 00.0
```

allows you to fine tune the M(ono) and S(tereo) levels of the incoming (front channel) source material, also known as M(id) and S(ide). The first knob controls the M level, the second knob controls the S, affecting the M/S ratio. **In most cases you will leave all of the input levels at 0, since K-Stereo does not depend on matrixing** to extract front channel ambience. M/S manipulation almost always results in a compromise, changing the original producer’s mix intentions, and the K-Stereo process gets around this problem with a totally different approach.

We provide the M and S levels as a convenient, powerful tool to fix problems. For example, you may have a recording with a weak, center-located vocalist. The only solution may be to turn up the center (M) level and/or turn down the S level. Try changing these in small, 0.1 dB steps to minimize this problem: As soon as you increase the M/S ratio, the mix changes---the stereo width suffers and the ambience goes to pot. (The bass and other center instruments also come up with the vocalist). *K-Stereo to the rescue!* That’s right, a touch of additional K-LEVEL can restore some of the sense of ambience and depth lost with the M/S ratio. It’s not perfect, but definitely a new and powerful *fix-it* tool. These levels are all remembered in the PRESETs, so by using MIDI automation, you can master an album which has one tune with weak vocal but the rest of the tunes are ok, with a seamless integration of sound quality from tune to tune.

Press INPUT LEVELS a second time to see this screen:

```
L 00.0    R 00.0    G  00.0
```

In this screen you can fine-tune the front input left and right balance, and/or reduce or raise the input gain. These levels are remembered in each individual preset. Ordinarily, leave G(ain) at 0 dB unless the output clips.

The Levels Screen is context-sensitive, and will return to M/S or L/R mode depending on the last way that you used it.

Frequently Asked Questions/Troubleshooting

FAQ

My Channel Balance seems to be off...what's the matter? Not to worry. There's nothing wrong with your K-Stereo unit. You must be trying to adjust channel balance with a steady-state test tone. If you try this with any standard reverb unit you will see it alternate in balance from left to right as you sweep a test tone up and down in frequency. This is perfectly normal. Since reverberation is random, at one specific frequency it may be left-heavy, at another frequency right-heavy. The proper way to check channel balance with a reverberator or with the K-Stereo box is to send mono music (or pink noise) signal and verify that **on the average** the sound is in the middle. A VU meter is best for this purpose instead of a peak meter, because random peaks may appear ambiguously on left or right, but the VU meter averages the aberrations.

What is K-Stereo? K-Stereo™ is a process that has been in secret development for over 10 years. K-Stereo extracts the ambience inherent in mixed recordings and spreads that uncorrelated ambience around the soundstage, while making the soundstage bigger. In addition, K-Stereo enhances the depth and imaging of the instruments and vocals.

What is K-Surround? K-Surround™ (available in Z-Systems Model z-K6) is the most effective method of extracting front channel ambience to the rear channels that we have ever heard. We find it is much more effective than the film mixer's trick of using a Dolby* Surround (Pro Logic) matrix decoder to feed ambience to the surround channels, which is sometimes called the "Magic Surround". Compare the two processes and you will find K-Surround sends a more natural and extended spread of the L/R ambience to the surround speakers, and "breaks down" less easily than the Dolby version. The effect is also program-dependent and should generally be reserved for music, but may also be used for voice. With voice, the natural ambience in the original source is brought to the rear, which may sometimes be useful for film work, when the original voice was recorded in a live space, it will not be necessary to use artificial reverberation to extend that live space to the surround speakers. *Dolby, TM Dolby Laboratories.

Can I use the K-Stereo Processor during mixdown? The answer is, absolutely—yes and no! Since K-Stereoization is designed to extract ambience from a source, it depends on how much ambience is already in your source(s). If you record a lot of dry, close-miked tracks, then the K-Stereo processor won't help much. But if you recorded a drum set, included a pair of room mikes panned left and right, bounced it to a pair of tracks, and then found during the final mix you'd like the drums more spacious and/or deep---K-Stereo to the rescue. K-Stereo will bring up the drum room mikes after the fact, with much better results and less mud than an artificial reverb chamber.

What does the "K" stand for? You may think of the K control as the level of the extraKted or reKovered ambience! But we had to give this new process a name, so might as well name it after the inventor....

How can I tell when I have enough K-Level? Our suggestion is to use a definitive monitor/reproduction system. Near-field monitors on top of a console do not qualify as "definitive". Why? Because near-fields often cause troublesome recordings that don't translate to the real world. The spacing of near-fields discourages a mix engineer from engineering much-needed spatiality and depth into a recording. Our second suggestion is to employ a stereo correlation meter on the front channels. Correlation meters read from +1 (pure mono) to 0 in the middle (random correlation) to -1 (stereo channels completely out of polarity). The sound is often more enjoyable when there is meaningful uncorrelated material in the recording. A general guide is that the correlation meter should approach 0, perhaps with random fluctuations a short way towards -1. But a lot of recordings sound better "tighter", and we don't listen with meters, do we? Finally, we suggest ear-training using acoustic recordings made with minimalist microphone techniques. The ear likes the warmth and support that natural ambience brings to a recording. Many multitrack recording engineers work hard to put that kind of spatiality in their recordings. Now K-Stereo makes that job a lot easier!

Why does my vocalist sound like he's recorded in a closet? That's because he **was** recorded in a closet! But seriously, K-Stereo cannot take a vocalist recorded in a resonant booth and convert that to Carnegie Hall; you will always hear that "room within a room" effect. Ambience extraction cannot distinguish between good and bad original ambience; bad skin still shows up under all that makeup. Thus, if the vocalist was recorded in a resonant booth and then reverb was added in the mix, when you use K-Stereo, you may detect a bit more of that resonance than you desire. Solutions include: Use less K-Stereo, try the HP filter, and/or try the parametric filter tuned to the resonance frequency. The good news is this is a rare problem, and usually the cure is so much better than the disease that it's easily tolerable.

Is there any way to reduce or get rid of extra reverberation in a recording? We've never heard of a *deverberator* , and when we've tried noise-gates, they only add an additional disease. The best solution we know is to *turn off the K-Stereo*, and try a high frequency rising shelf or presence filter to try to increase the clarity of the direct sound. Possibly using a sibilance controller or dynamic equalizer to keep the effect from becoming aggravating at high levels. If you can do all this without making the sound harsh or unpleasant, then great! Good luck!

SPECIFICATIONS

AES/EBU standard digital inputs and outputs at all standard (fixed) sample rates between 44.1 and 96 kS/s. Output wordlength adjustable to 24-bits (TPDF dithered), 20 bits (TPDF dithered), or 16 bits (POW-R™ Dither). 16-bit dither is not available at rates above 48 kS/s. Model **DD-2** is 2-channel in and out via 1 AES/EBU connection.

PATCHING RECOMMENDATIONS. We recommend patching this processor early in the digital signal chain, in front of any compressors or other dynamics processors. Patch the Model DD-2 between a 2-channel reproducer and a digital console, or in a console's digital insert point. You can also use an external A/D/A converter around the K-Stereo box for use with an analog system. Use the highest quality A/D/A for this purpose (ideally, 96 kHz/24 bit).

With Noise-Reduction processors: We suggest patching noise-reduction processors in front of the KS-processor, so that you can restore ambience lost during noise reduction.

PROCESSING: Ambience recovery is performed using unique psychoacoustically-derived K-Stereo™ algorithms (patent pending) developed by Bob Katz. Equalization uses low-distortion filtering algorithms developed by Glenn Zelniker. Dithering uses either TPDF (flat) dither at 24 or 20 bits, or the highly-transparent POW-R dither at 16 bits, developed by the POW-R consortium. **All processing is performed using floating point calculations, so there is no possibility of internal overload as long as the output level does not overload.**

DOWNMIX-COMPATIBILITY. Mono compatibility is good to excellent. Some algorithms have better downmix compatibility than others and it is also dependent on the source material. As you would during any mastering session, always test for downmix compatibility. In general, you will be very pleased.

BYPASS CONDITIONS. BYPASS produces a true bit-for-bit clone of the digital source. Turning all processing off and setting the wordlength to **24 bits** is equivalent to
BYPASS.

Model DD-2 Block Diagram