MODEL 166A

PROFESSIONAL
DYNAMICS PROCESSOR

dbx®

OPERATION MANUAL

CAUTION

RISK OF ELECTRIC SHOCK
DO NOT OPEN

ATTENTION: RISQUE DE CHOC ELECTRIQUE - NE PAS OUVRIR

WARNING: TO REDUCE THE RISK OF FIRE OR ELECTRIC SHOCK, DO NOT EXPOSE THIS EQUIPMENT TO RAIN OR MOISTURE.

CAUTION: TO REDUCE THE RISK OF FIRE OR ELECTRICAL SHOCK, DO NOT REMOVE COVER (OR BACK). NO USER SERVICABLE PARTS INSIDE. REFER SERVICING TO QUALIFIED SERVICE PERSONNEL.

This symbol, where ever it appears, alerts you to the presence of uninsulated dangerous voltage inside the enclosure - voltage that may be sufficient to constitute a risk of shock.

This symbol, wherever it appears, alterts you to important operating and maintenance instructions in the accompanying literature. Read the manual.

This manual is part number 18-2012-C

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A Harman International Company

INTRODUCTION

Congratulations on choosing the dbx 166A Professional Dynamics Processor. The 166A provides two channels of noise gating, OverEasy® or classic Hard Knee compression and PeakStop® limiting to give complete control of signal dynamics to studios, sound reinforcement companies, musicians, or anyone who needs quality processing quickly and easily. We recommend that you take a moment and read through the manual as it provides information that will assist you in using your unit to its fullest potential. Features Include:

- Stereo or Dual Mono operation of gating, compression and peak limiting.
- OverEasy®/Hard Knee Selection - allows selection between our famous OverEasy compression curve and the classic "Hard Knee" curve popularized by the original dbx 160, 161 and 162.
- Expander/Gate Circuit - with variable release time and +15dB maximum threshold.
- Compressor Mode switch - selects from two sets of program-dependent Attack and Release times to tailor response for individual instruments or mixed program material.
- Selectable Low Frequency Shelf (via Contour button) in the Sidechain Path - recommended when compressing mixed program material to prevent low frequency energy from punching "holes" in the sound.
- PeakStop® Limiting - provides control of maximum peak levels at the output of the 166A regardless of any other control. PeakStop comes after the compression, gating and other circuitry including the output gain, so it lets an absolute limit be put on the peak excursions before they reach the output.
- True RMS Level Detection - senses the power in the program in a musical manner, much as human hearing does, giving results superior to peak or average detection.
- Hardwire System Bypass Buttons on both channels - allow the audio to pass even if the unit is unplugged, and are also useful for comparing the processed and unprocessed signal.
- 10-Segment LED Display for GAIN REDUCTION (up to 30dB).
- Electronically Balanced XLR and 1/4" TRS Input and Output Jacks
- Separate Sidechain Inserts - enables an outboard processor or signal to control compression or gating.
- DC-Controlled Parameters - the signal does not pass thru any of the parameter controls. Instead a DC voltage controls all functions; this eliminates any possibility of potentiometer noise developing over time.

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INSPECTION

Verify that the 166A package contains the following:

- ○ 166A Unit
- ○ AC Power Cord
- ○ Operation Manual
- ○ Registration Card

QUICK SETUP

To get your unit up and running as quickly as possible, do the following steps. For more detailed information, refer to the specified pages.

○ Unpack and Inspect the 166A Package. Page 3
○ Connect the 166A to Your System. Page 15
○ Set Levels and Controls as Needed. Page 4

WARRANTY

This warranty is valid only for the original purchaser and only in the United States. We warrant dbx products against defects in material or workmanship for a period of two years from the date of original purchase for use, and agree to repair or, at our option, replace any defective item, except external power transformers, without charge for either parts or labor.

IMPORTANT: This warranty does not cover damage resulting from accident, misuse or abuse, lack of reasonable care, the affixing of any attachment not provided with the product, loss of parts, or connecting the product to any but the specified receptacles. This warranty is void unless service or repairs are performed by an authorized service center. No responsibility is assumed for any special, incidental or consequential damages. However, the limitation of any right or remedy shall not be effective where such is prohibited or restricted by law.

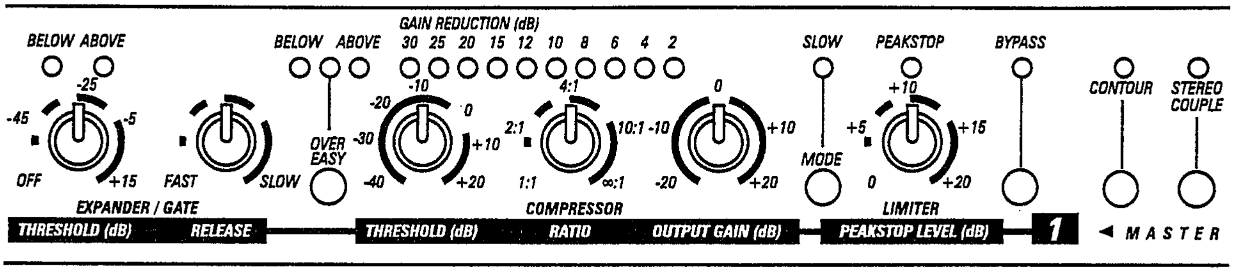
Simply take or ship your dbx product prepaid to our service department. Be sure to include your sales slip as proof of purchase date. (We will not repair transit damage under the no-charge terms of this warranty.) dbx will pay return shipping.

NOTE: No other warranty, written or oral is authorized for dbx products.

This warranty gives you specific legal rights, and you may also have other rights which vary from state to state. Some states do not allow the exclusion of limitations of incidental or consequential damages or limitations on how long an implied warranty lasts, so the above exclusion and limitations may not apply to you.

OPERATING CONTROLS

Front Panel



BYPASS Button and LED: Depress this button to create a "hard-wire bypass" of the 166A's circuitry (i.e., unaltered input signal will pass through the unit even if it is unplugged). Note that BYPASS works independently for each channel, even when the unit is stereo-coupled (via the STEREO COUPLE button).

In Bypass mode, the input is sent directly to the output, bypassing the 166A's processing and controls and presenting unaltered input signal at the 166A's OUTPUT. Bypass mode is especially useful for making comparisons between processed and unprocessed signals.

The BYPASS LED turns On in Bypass mode if the 166A is being provided with AC power.

GAIN REDUCTION Meter: This meter displays how much the signal is being attenuated by the compressor, the gate, or both.

EXPANDER/GATE Section

Expander/Gate THRESHOLD Control and LEDs (BELOW/ABOVE): Adjusting this control sets the level at which the gate will open and allow the signal at the input to pass through to the output. Turning the knob fully counterclockwise (to OFF) allows the gate to pass signals unattenuated, effectively bypassing the gate. Turning the knob fully clockwise causes the gate to attenuate input signals below +15dBu.

The two Expander/Gate LEDs indicate the relationship of the input signal level to the threshold setting. The red LED lights when the signal is BELOW threshold, the green LED lights when the signal is ABOVE threshold.

Note: The 166A's expander/gate attack rate (which controls how fast the signal is restored after being attenuated) is internally set to be very fast - fast enough to allow all of the transient at the beginning of a note, vocal or spoken word to come through.

Note: The 166A's expansion ratio is internally fixed, at approximately 10:1. This ratio helps to eliminate the artifacts normally associated with common switch gates. Attenuation is >50dB.

Expander/Gate RELEASE Control: This control determines the rate at which the gate closes once the signal at the INPUT or SIDECHAIN INSERT falls below the threshold. SLOW settings are useful for gating out noise present behind vocals and acoustic instruments. FAST settings are useful for tightening up the sound of percussion (e.g., kick or snare drum) and drying up leakage from other instruments into percussion tracks.

Note: The gate release rate is “accelerating” in that the dB/Sec rate continually increases as the gate closes.

COMPRESSOR Section

OVEREASY Button: Depress this button to select the OverEasy® compression characteristic. The amber THRESHOLD LED turns On when the signal is in the OverEasy region. When this button is in the Out position, the 166A operates as a hard knee compressor/limiter. (Amber OverEasy LED is active only in OverEasy Mode.)

In Hard Knee mode, the threshold of compression is defined as that point above which the output level no longer changes on a 1:1 basis with changes in the input level. See Figure 1.

In OverEasy mode, the threshold of compression is defined as the middle of the OverEasy threshold region, that is, “half-way” into compression, as shown in Figure 2.

Compressor THRESHOLD LEDs: These three LEDs indicate the relationship of the input signal level to the threshold of compression. The green BELOW LED is On when the signal is below threshold and the red ABOVE LED is On when the signal is above threshold. When the 166A is switched to OverEasy mode, the amber LED is On when the signal is in the OverEasy region (See Figure 2).

Note: Even though no input signal is being applied, it is normal for the LEDs to flicker when the power is applied or removed.

Compressor THRESHOLD Control: Adjust this knob to set the threshold of compression from -40dBu (7.8mVrms) to +20dBu (7.8Vrms). Setting the Compressor THRESHOLD control to +20dB will prevent all but the highest level peaks from being compressed. (Setting the Compressor RATIO to 1:1 will turn the Compressor off, regardless of the setting of the Compressor THRESHOLD control.)

In Hard Knee mode (OVEREASY button out), the THRESHOLD sets a reference level above which input signals will be processed by the 166A’s gain change circuitry in the manner defined by the setting of the RATIO control. Input signals which fall below this level will pass through the 166A unprocessed (except for fixed gain changes directed by the OUTPUT GAIN control). See Figure 1.

In OverEasy mode (OVEREASY button depressed), signals begin to gradually activate the 166A’s gain change circuitry as they approach the THRESHOLD reference level and they do not get fully processed in the manner defined by the RATIO control until they have passed somewhat above the THRESHOLD reference level. In OverEasy mode there is no distinct point at which processing begins, and the THRESHOLD setting corresponds to a point on the input/output transfer curve midway between the onset of processing and that point at which the transfer curve corresponds to the setting of the RATIO control. Figure 2 shows the OverEasy compression curves and how they correlate with the THRESHOLD LEDs.

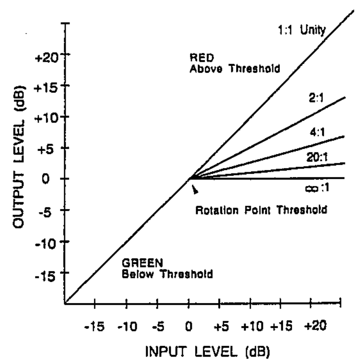


Figure 1: Hard Knee Compression Curve and Threshold LEDs

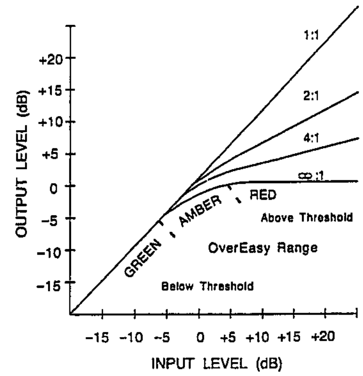


Figure 2: Over Easy Compression Curve and Threshold LEDs

Compressor RATIO Control: Rotate this control clockwise to increase the amount of compression from 1:1 (no compression) up to Infinity:1 (no increase in output level, regardless of input level increases above threshold).

When an input signal is above the THRESHOLD reference level, the setting of this control determines the number of decibels by which the input signal must change in level to produce a 1dB increase in the signal level at the output of the 166A. A setting of 2:1 indicates an input:output ratio wherein a 2dB increase in signal (above threshold) will produce a 1dB increase in output signal. A setting of 2:1 indicates that an infinite increase in input level would be required to raise the output level by 1dB. In other words, the output level stays constant when the input signal rises above threshold.

OUTPUT GAIN Control: Adjust this control to vary the amount of fixed gain (up to ±20dB) in the 166A's output amplifier stage. The OUTPUT GAIN control does not interact with the threshold of compression.

The OUTPUT GAIN control is especially useful to compensate for the RMS level decrease which results from the 166A's dynamic processing effects. After you adjust the 166A's controls for the desired amount of compression (and gating), set the OUTPUT GAIN to add the same amount of gain that is shown on the GAIN REDUCTION meters. For example, if the average amount of gain reduction shown on the meters is 10dB, then setting the OUTPUT GAIN control to 10dB will compensate for the +10dB average level reduction at the output. Note that the GAIN control comes before the PeakStop circuit.

Note: Because +20dB of gain can be added at the 166A output, it is possible to cause clipping even when the input level is within the specified range. For example, when the COMPRESSION RATIO is set at a low number, extreme clockwise rotation of the OUTPUT GAIN may cause the 166A output stage to clip program peaks. Therefore, for normal operation we suggest beginning with the OUTPUT GAIN set at "0dB" (12 o'clock position). Where the circuit fed by the 166A has a high input sensitivity, lowering the 166A's OUTPUT GAIN setting can avoid the need for an attenuation pad.

Compressor MODE Button and SLOW LED: This button selects from two sets of program-dependent attack and release times to tailor response for individual instruments or mixed program material. Pressing the button In activates SLOW compression and the SLOW LED lights. When the button is Out, the rate of compression is faster.

For the smoothest compression, press In the MODE and OVEREASY buttons. For the most aggressive compression leave both MODE and OVEREASY buttons Out (i.e., for a faster Hard Knee compression).

LIMITER Section

PEAKSTOP LEVEL Control and LED: This control allows you to set the maximum peak output level of the 166A regardless of any other control. PeakStop comes after the compression, gating and output gain circuitry; this provides for an absolute limit to be put on the peak excursions at the output. PeakStop works instantaneously; you can apply moderate amounts of dbx's OverEasy compression and still be protected from large transients, other short-term overloads and overmodulation.

PeakStop is a smooth well-controlled soft clipper whose behavior is sonically similar to the gentleness of OverEasy compression; its clipping is much preferable to a power amp's or analog-to-digital converter's. PeakStop rounds the corners of a peak rather than cutting it off sharply. By making a signal's leading and trailing edges curved instead of sharply angled, it reduces the amount of higher odd-order, offensive-sounding harmonics that conventional hard clipping causes.

The level at which PEAKSTOP is activated is adjustable from +0dB to +20dB. Note that small signal excursions above the set value of PEAKSTOP are possible, to allow the rounding to occur. Therefore, for applications where you must not exceed a given ceiling, set the PEAKSTOP control 1dB to 2dB below the ceiling.

The PEAKSTOP LED illuminates whenever peaks attempt to exceed PeakStop level and are reduced in amplitude. If the PeakStop LED illuminates when the PEAKSTOP LEVEL control is set to +20dB, the headroom capabilities of the 166A are being exceeded and hard clipping is occurring.

MASTER Section

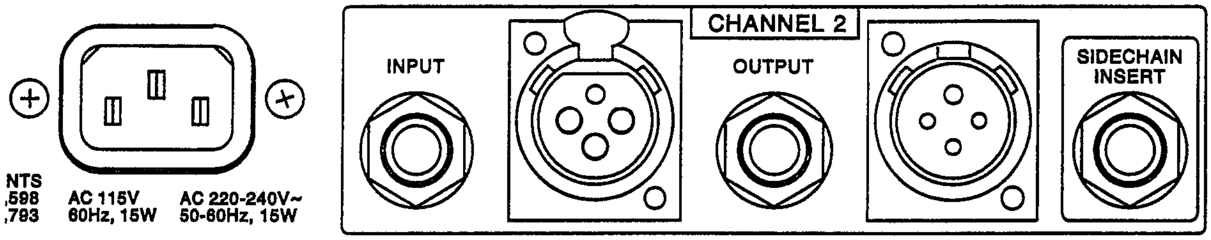
CONTOUR Button and LED: Depress this button to make the 166A's detection circuitry less sensitive to low frequency energy, preventing this energy from punching "holes" in the sound, especially with mixed program. With the CONTOUR button Out, the 166A's detector is frequency-independent. The CONTOUR LED turns On when the CONTOUR button is depressed.

STEREO COUPLE Button and LED: This button toggles the unit between stereo and dual mono operation. Press the STEREO COUPLE button In for stereo operation where Channel 1 becomes the master controller for both channels. All of Channel 2's controls, buttons, and LEDs will be disabled (except for Channel 2's BYPASS button, and its GAIN REDUCTION LEDs), since Channel 2 is the "slave." Note that the detection circuitry senses the true RMS levels of the combined signal, so it is unaffected by phase shifts (or other discrepancies) between the channels. This ensures stereo compression without loss of imaging stability.

With the STEREO COUPLE button Out, the unit functions as two separate mono compressor/gates, each with its own independent controls.

The STEREO COUPLE LED indicates that the 166A is stereo-coupled.

Rear Panel



INPUT (BALANCED) Jacks: The Tip/Ring/Sleeve phone jack and XLR-type jack are wired in parallel; either INPUT will accept an audio signal for processing by the 166A. The phone jack accepts a standard TRS 1/4" phone plug for a balanced input source, or a 2-circuit (Tip/Sleeve) 1/4" phone plug for an unbalanced source. The XLR-type jack is wired pin 2 HOT (+), pin 3 COLD (-) and pin 1 GROUND.

Note: A given pair of channel input jacks (e.g., Channel 1 XLR INPUT and Channel 1 1/4" INPUT) are internally connected (TIP = Pin 2, RING = Pin 3, SLEEVE = Pin 1). Therefore, if one of the jacks is unbalanced, then the other jack will be unbalanced. For example, if a 1/4" INPUT jack is used with a mono cable, and is therefore unbalanced, the XLR INPUT jack will also be unbalanced (Pin 3 shorted to ground).

OUTPUT (BALANCED) Jacks: The Tip/Ring/Sleeve phone jack and XLR-type jack are wired in parallel; either OUTPUT will send an audio signal on to a load. The phone jack accepts a standard TRS 1/4" phone plug for a balanced output load, or a 2-circuit (Tip/Sleeve) 1/4" phone plug for an unbalanced load. The XLR-type jack is wired pin 2 HOT(+), pin 3 COLD (-) and pin 1 GROUND. For proper unbalanced operation, the unused pin (either pin 2 or 3) must be grounded. Nominal output signal level is +4dBu into 600Ω, and typical maximum output level is +20dBu into 600Ω.

Note: A given pair of channel output jacks (e.g., Channel 1 XLR OUTPUT and Channel 1 1/4" OUTPUT) are internally connected (TIP = Pin 2, RING = Pin 3, SLEEVE = Pin 1) and can simultaneously deliver the same signal to two separate loads, but if one of the jacks is unbalanced, then the other jack will be unbalanced. For example, if a 1/4" OUTPUT jack is used with a mono cable, and is therefore unbalanced, the XLR OUTPUT jack will also be unbalanced (Pin 3 shorted to ground).

SIDECHAIN INSERT Jack: This jack accepts a standard TRS 1/4" phone plug and provides a connection to the 166A detector path. The RING acts as a Send, carrying a buffered version of the signal present at the 166A INPUT jack, at an impedance of 2kΩ. The TIP acts as a Return for equipment to feed the 166A's detector circuitry, such as an equalizer for de-essing or frequency-sensitive gating/compression. You can also drive the 166A SIDECHAIN INSERT with the output of most equipment, by using a 1/4" mono phone plug. Input Impedance is greater than 10kΩ.

Note: When a cable is plugged into this jack, it automatically breaks the connection from the INPUT jack to the 166A's detection circuitry.

AC Power Jack: This jack accepts an IEC-type power cord (as shipped with the unit). Plug the cord into the unit and mains power. Note that the 166A does not have a power switch. It is recommended that the 166A be "On" at all times. Power consumption is low. If you do not plan to use the 166A for an extended period of time, unplug it.

Warning: Be sure to verify both your actual line voltage and the voltage for which your Model 166A was wired, as indicated on the rear panel of your unit. Connection to an inappropriate power source may result in extensive damage which is not covered by the warranty.

Operating Notes

Expander/Gate Applications

Note: Control settings for each application are suggested as a starting point. Adjust them for your requirements.

Gating Dry Percussive Sounds (e.g., Snare Drum, Kick Drum)

To effectively gate percussive sounds with a high level transient, you need to set the 166A's gate controls to ensure that the gate is less sensitive to nearby signals that would cause the gate to open or "false trigger."

Set the RELEASE setting fast enough to enable the gate to close very quickly once the signal falls under the THRESHOLD. The RELEASE can also be used to shape the envelope of the sound.

Note: Fast gating of sustained low frequency signals can result in "chattering." Because the 166A is capable of extremely fast gating, make sure the RELEASE time is longer than one full cycle of the gated signal's fundamental frequency. To eliminate any "chattering," simply adjust the RELEASE time to a longer time (slower rate). The proper THRESHOLD setting will also minimize false triggering and "chattering."

These type of settings are most useful for tightening up drum tracks, removing the "ring" from some drums, or gating out the leakage of one drum through another's mic.

Gating Sounds That Have Longer Decay (e.g., Cymbal, Piano)

To effectively gate sounds which have more decay after the initial transient, set the RELEASE control slow enough to allow the gate to remain open and capture the signal's entire envelope.

The gate can also be used to "dry up" a track or mix that has too much reverb or ambience. Set the RELEASE control so that the natural decay of the sound is somewhat truncated.

Changing Sound Quality

The 166A's gate can effectively change the sonic character of a sound because it can reduce or otherwise alter the quality of instrumental ambience and reverb. For example, as an instrument stops, its reverberation level will fall through the 166A's THRESHOLD setting. It can now be made to die out more quickly - faster than the natural delay (of the sound). Experiment with different THRESHOLD and RELEASE settings to change the "tail" of the sound; a FAST RELEASE setting will nearly eliminate reverb.

Keyed Gating

Keyed gating, that is, controlling the gating of one signal by another, can be used to add dynamics to a sound (e.g., creating perfectly in-synch playing and overdubbing among individual instruments or "fattening" a dynamically weak track).

To create two distinct channels of bass guitar for your mix (by splitting the bass signal into two channels and synchronizing one channel of bass guitar with the kick drum), start by feeding one channel of bass directly into the mix and the other into the gate's INPUT. Then key the gate with a signal from the kick drum (connected to the SIDECHAIN INSERT - adjust controls as needed). The gated bass track will now open with each kick, adding punch and dynamics. This can really tighten up the tracks and add life to the mix.

Another example of keyed gating is using the drum signal to key an oscillator which is set to an appropriate frequency to "tune" and "punch up" the drum sound.

> **Note:** For all keyed gating applications, be aware to adjust the compressor accordingly or bypass it by setting the Compressor RATIO fully counterclockwise to 1:1

Frequency-Sensitive Gating

Frequency-sensitive gating lets you use the SIDECHAIN INSERT to tune the response of the gating action. For example, if you're gating a kick drum in a track with lots of leakage, you can tune in to the frequency of the kick with an outboard EQ and the gate will respond only to that drum. Feed the kick drum signal both directly into the gate and also through an equalizer which is connected to the SIDECHAIN INSERT. With the equalizer adjusted so that only the desired signal is strong at the SIDECHAIN INSERT, the gate becomes even more selective in opening.

Basic Compressor Applications

Note: Control settings for each application are suggested as a starting point. Adjust them for your requirements. In general, the "smoothest" compression is achieved with the OVEREASY and MODE (Slow) buttons In, while the most "aggressive" compression is achieved with a Hard Knee fast setting (i.e., OVEREASY and MODE buttons Out).

To compress a mix, begin with a low RATIO setting, THRESHOLD set for a few dB of Gain Reduction, and SLOW MODE, OVEREASY and CONTOUR buttons In.

Smoothing Out Variations in Microphone Levels

Variations in signal level can occur when the distance between a vocalist and a mic changes, or when the dynamics of a voice changes relative to a vocalist's range. To smooth out these variations, start with the 166A in OverEasy SLOW mode (both the OVEREASY and MODE buttons In) and adjusted for a low to medium compression RATIO (e.g., 4:1). Adjust the THRESHOLD control so that the GAIN REDUCTION meters show 6dB to 10dB of gain reduction, then increase the RATIO if necessary. Due to the gentle OverEasy characteristic of your 166A you will find that even fairly high ratios are handled transparently. If the lower energy of the vocals is compressed too much (e.g., if the voice sounds too thin or its lower register presence is lost), press the CONTOUR button In to allow more of the original low energy to pass through the 166A unaffected.

Smoothing Out Variations (and Increasing Sustain) in Musical Instrument Levels (e.g., Bass Guitar, Electric Guitar, Synthesizer)

To achieve a smoother electric (or electronic) bass sound, compress the instrument's output with a RATIO of approximately 4:1, then adjust the THRESHOLD control for 10dB to 12dB of gain reduction. Compression lessens the loudness variations among the strings and increases the bass' inherent sustain. Other instruments, such as horns, vary in loudness depending on the note being played, and benefit similarly. Note that if the compressed bass sounds smooth, but too thin for your needs, try pressing In the CONTOUR button to thicken the sound.

To control untimely volume shifts in "hot" guitar or synth parts and to keep them from overloading your tape deck or mixer during recording and live performances, start with a slow Hard Knee compression (MODE button In, OVEREASY Out), the RATIO at approximately 5:1 and the THRESHOLD set to the average maximum level of the track - this will ensure that only the offending "hot" part is compressed. Use CONTOUR, if necessary.

To add sustain to guitar or synthesizer string sounds, begin with a higher RATIO (from 10:1 to ∞:1), then adjust the THRESHOLD control to taste. For example, to alter the envelope of a synthesizer sound that has a bite on its attack, but ends with a long release time, begin by playing slow, but steady, synth stabs or chords, while compressing the sound heavily (with a higher RATIO). After adjusting the THRESHOLD to taste, experiment with different combinations of MODE and OverEasy settings (e.g., FAST MODE with OverEasy In, or SLOW MODE with Hard Knee, etc.) to hear the effect each mode has on the attack and release of the envelope. Heavy compression of guitars and synths, as they are being recorded to digital formats, can often help revive their sense of "analog life."

Fattening Kick Drums and Compressing Other Drums

Weak, flabby kick drums often have too much boom, and not enough slap. To tighten them up, start with the 166A adjusted for a medium to high RATIO (e.g., 6:1), adjust the THRESHOLD control so that the GAIN REDUCTION meters show 15dB of gain reduction, then increase the RATIO if necessary. In OverEasy mode, the 166A takes slightly longer to react than in Hard Knee mode, and will therefore emphasize the slap at the beginning of the note and reduce the boominess of its body. The 166A also works well for tightening snare drums and tom toms and can be used with drum machines to effectively alter the character of any electronic drum sound.

Cymbals and tom-toms can be effectively compressed (using the 166A's Sidechain) to help prevent tape saturation. Split the drum signal, sending one channel directly to the 166A's INPUT and the other channel to an equalizer (e.g., dbx's 242 Parametric Equalizer or the 30 Series Graphic Equalizers). Then connect the equalizer's output to the 166A's SIDECHAIN INSERT. The equalizer can be adjusted for boost with a peak of about 5kHz, causing the cymbal to be compressed on a very loud crash, stopping tape saturation at high frequencies, where there is less headroom. However, gentle tapping of a drumstick or brushing of the cymbal will not be affected. Assuming the tom-tom is a lower frequency instrument and can be better tolerated by the tape, it has less need for compression. Equalization in the Sidechain circuit means that the compressor is not triggered as readily by a loud tom-tom beat as by an equally loud cymbal crash. Refer to the next page for more Sidechain applications.

For drum kit submixes (e.g., mixing multiple drum tracks to two tracks while using both channels of a 166A for compression), consider backing off the RATIO on each channel (down to 2:1) to avoid an excess of cymbal "splattering." In larger multitracking systems, compress the kick and snare separately. A further possibility is to heavily compress a stereo submix of toms and leave the remaining percussives unaffected.

Raising a Signal Out of a Mix

Since reducing dynamic range increases the average signal level by a small amount, a single track can be raised out of a mix by boosting its level slightly and applying compression. Start with a 2:1 RATIO and a relatively low THRESHOLD setting (-20dB). Adjust both controls as necessary.
Compressors have also been used to bring vocals to the forefront of a mix in volume-restricted studios (e.g. home studios). Start by adding a foam windscreen to the mic (if it doesn't have one). Set the RATIO to 10:1 and the THRESHOLD to -10dB. With your mouth approximately 2 inches from the mic, sing the vocal part, but with less volume than normal. Use phrasing to give the part some intensity. An equalizer (e.g., a dbx 242 Parametric Equalizer, dbx 30 Series Graphic Equalizers) or a vocal effects device (e.g., reverb, delay, distortion) can be added to further define the performance.

It is also possible to separate certain vocals or instruments from a mono program already mixed: refer to frequency-weighted compression on page 9.

> **Note:** When compressing a stereo program with a 166A, the factors affecting a compression curve and the actual RATIO and THRESHOLD settings, are like those previously covered with reference to single channels of program material. However, it will generally be found that large amounts of compression are more audible in a mixed stereo program than they might be on the separate tracks that were mixed to create the program.

Preventing Tape Saturation

With programs of widely varying levels, compression can prevent recording levels (e.g., cymbal tracks in a final mix or drum kit submix) from saturating tape tracks (see frequency-weighted compression, below).

Preventing Digital Overload

Some digital recorders and samplers produce audible distortion when they exceed their headroom (i.e., the range above their maximum operating level). The 166A effectively ensures that audio input does not overload a digital recorder's A/D (analog-to-digital) converters. The 166A can perform this function quietly enough for all digital media. To use the 166A so that no changes in gain occur unless an emergency arises (wildly excessive levels), set Hard Knee mode On, the RATIO to ∞:1, and the THRESHOLD to the highest permissible level.

> **Note:** PeakStop limiting can also be used to prevent raucous-sounding digital overload.

Speaker Protection (Auditoriums, Churches, Mobile DJs and Sound Systems)

Compressors are frequently used to prevent excessive program levels from distorting power amps and/or damaging drivers in a sound-reinforcement system (whether you're doing auditorium, church, or club sound engineering, or are a mobile DJ, or like to push the limits of your home's audio entertainment center). Set the 166A for limiting (Hard Knee mode On, with a RATIO of 10:1 or greater) and adjust the THRESHOLD to provide 15dB or more of compression (just a few dB below the input clip). For low-level signals, the 166A won't change gain, but if large signals come along, the gain will be reduced to prevent clipping and save sensitive system components from excessive heat buildup or other type of damage.

In circumstances where the 166A is expected to cause no change in gain unless an emergency arises (wildly excessive levels), some operators set Hard Knee mode On, the RATIO to ∞:1, and the THRESHOLD to the highest permissible level. As with preventing digital overload, the 166A's PeakStop limiter can be used instead of or in combination with the 166A's compression.

As a general rule, compressors should be as close to the amplifiers as possible in the signal chain. If the 166A is placed before the EQ (equalizer), for example, a potentially damaging boost in the EQ won't be seen by the 166A and the speakers may be damaged. (see Multi-way speaker systems, page 10). For maximum sound pressure levels, large sound reinforcement systems frequently use a separate compressor on each output of the electronic crossover(s). For a stereo sound-reinforcement system, one 166A can be used for each stereo band (low, low-mid, mid, etc.).

Raising Average Level in PA Systems

Limiting (i.e., compression at high ratios like ∞:1) also benefits intelligibility by allowing low-level input signals to be reproduced through the system at higher volume. In a musical performance, this provides additional intimacy so that a vocalist's whispers are heard more clearly. The OverEasy curve available with the 166A permits a very high amount of compression (RATIO of 10:1 or greater) to be used in many situations. This allows dynamic speakers, vocalists and other musicians to concentrate on their presentation or performance without worrying about the ill effects of volume changes.

Using Your EQ to Reduce Feedback in Live Settings (Indoor and Outdoor Concerts, Churches)

You can use your 166A and EQ (equalizer) to reduce feedback in clubs, churches, outdoor concerts and other live settings. Patch or insert the 166A into the main output of a mixer, set the 166A to Hard Knee mode and slowly increase OUTPUT GAIN until the first feedback "ring" occurs, then set up the 166A with its RATIO at ∞:1 and THRESHOLD low. The 166A will catch the first feedback ring and hold it as a constant tone so you can adjust your EQ to minimize it. Continue to increase your console gain and set your EQ until the next 3 or 4 "ring" frequencies have been compensated for.

The 166A as a Line Amplifier

To use the 166A as a line amplifier, adjust the RATIO control fully counterclockwise (1:1 position), THRESHOLD fully clockwise (+20), PeakStop to +20 and OUTPUT GAIN to whatever setting is required for the application. Remember, excessive gain may lead to output clipping of high level signals. To add compression, adjust the RATIO and the THRESHOLD controls to the desired settings.

Frequency-Weighted Compression (Sidechain Application)

It is possible to separate certain vocals and instruments from a mix by frequency-weighted compression. With an equalizer (such as a dbx 242, or a dbx 30 Series EQ) inline ahead of the SIDECHAIN INSERT (but not in the audio path), the equalization settings do not shift the timbre or frequency response of the audio signal. They merely alter the threshold response of the compressor on a "frequency-weighted" basis.

With this arrangement, raising certain frequencies on the equalizer causes them to be suppressed in the audio signal. A relatively high THRESHOLD setting can allow normal sounds to be unaffected while solo and very loud sounds are compressed. (Of course, when compression occurs, the level of the entire program is affected - however, if the 166A's CONTOUR button is pressed in, even more of the signal's lower energy can be preserved.) Depending on the THRESHOLD setting, lower amplitude fundamentals or harmonics will not cause compression, and the program is not subject to the phase shift normally caused by

program equalization.
When recording cymbals and tom-toms, a compressor with an equalizer in the Sidechain path can help prevent analog tape saturation. The equalizer can be adjusted for boost with a peak of about 5kHz, causing the cymbal to be compressed on a very loud crash, stopping tape saturation at high frequencies, where there is less headroom. However, gentle tapping of a drumstick or brushing of the cymbal will not be affected. Assuming the tom-tom is a lower frequency instrument and can be better tolerated by the tape, it has less need for compression. Equalization in the Sidechain circuit means that the compressor is not triggered as readily by a loud tom-tom beat as by an equally loud cymbal crash.

The converse of the above EQ technique may be used: dipping the equalizer bands causes any sound with dominant energy in the affected register to pull the level up because the 166A will detect a need for less compression.

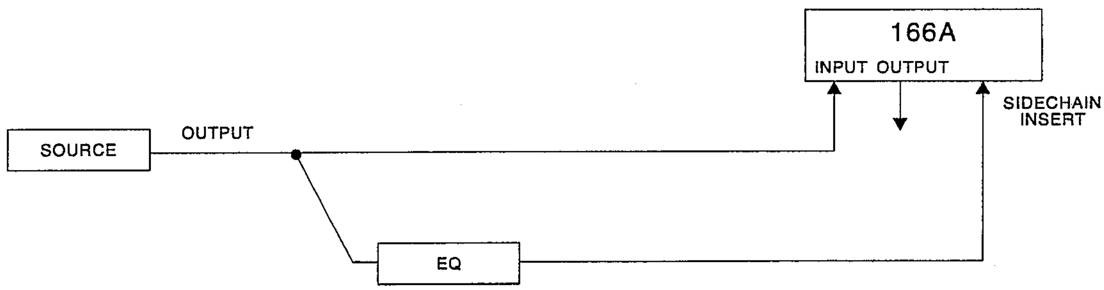


Figure 3: Frequency-Weighted Compression

De-Essing

To apply de-essing to vocals (i.e., a reduction of sibilance), use a parametric equalizer (e.g., a dbx 242) in the SIDECHAIN circuit and set it to boost the specific frequency range where the vocal "hiss" or lisp occurs (generally in the 4kHz - 6kHz region). This pre-emphasizes the already "hissy" vocal input to the Sidechain. Used in conjunction with a moderate to high THRESHOLD and RATIO, and a fast MODE setting, this arrangement greatly attenuates the "essing" without affecting the basic sound quality or balance of the voice. While it is true that all frequencies are lowered in level when the compressor is triggered, generally the "sss" sound occurs alone, before or after the dominant tone in the voice.

Increasing Sustain

To increase the sustain of a musical instrument (e.g., a guitar or bass), use an equalizer in the level Sidechain circuit and boost the EQ in the dominant frequency range of the instrument. Set the 166A for slow Hard Knee compression (MODE button In, OverEasy button Out), with a fairly low THRESHOLD and a moderate RATIO.

Multi-Way Speaker Systems

If a single compressor is to be used with a multi-way speaker system (i.e., before the crossover, after the EQ), the system operator is faced with the problem of keeping levels below the point of damage of the most sensitive part of the system. If, for example, mid-range drivers are frequently damaged, the whole system must be operated at a lower sound-pressure level, or additional mid-range drivers must be added. By inserting an equalizer in the Sidechain path to the 166A, it can be made more sensitive to frequencies in the range handled by the sensitive drivers. The system can then be run at higher levels and will only be dropped back when damaging signals are present.

Using a Filter in the Sidechain Circuit

The results of inserting a filter in the Sidechain circuit are basically the same as obtained with an equalizer, as previously described. Those frequencies passed by the filter are subject to compression (or at least they are subject to considerably more compression than those frequencies outside the passband). Because a passive filter can have insertion loss, it may be necessary to lower the 166A's THRESHOLD setting to maintain a given amount of gain reduction within the filter passband; this can be determined by monitoring the 166A's THRESHOLD LEDs.

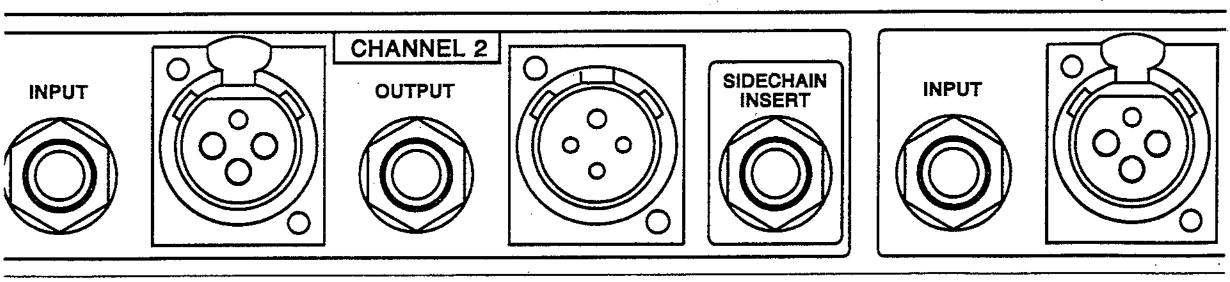
Pre-Emphasis for Broadcast Applications

By inserting a pre-emphasis filter network in the Sidechain path of a 166A processing pre-emphasized audio, higher levels can be run within the headroom limitations of the broadcast chain.

Connecting The 166A To Your System

Basic Connection

The 166A has balanced inputs and outputs, and can be used with any line-level device. Some common examples include: mixing consoles, musical instruments, patch bays and other signal processors. For more specific cabling information, refer to Installation Considerations, page 13.



For all connections, refer to the following steps:

1. **Turn Off all equipment before making any connections.**

2. **Mount the 166A in a 1U rack space (Optional).**

   The 166A requires one rack space (height) and 1 rack space (width). It can be mounted above or below anything that doesn't generate heat, since it requires no special ventilation. Ambient temperatures should not exceed 113°F when equipment is powered.

   Note: Avoid over-tightening of rackmounting screws as this could damage the front panel.

   Caution: Never remove the cover. There are no user-serviceable parts inside, and you run the risk of an electric shock.

3. Make connections via XLR or 1/4" TRS jacks according to your requirements.

Typical patch points include: a mixer's channel or subgroup inserts when using the 166A on individual instruments or tracks; the mixer's main outputs or bus inserts when mixing; an instrument preamp's effects loop when using the 166A for guitar or bass; main outs of a submixer (e.g., keyboard mixer) as the signal is sent to main mixer; between a DAT's output and an analog cassette input. When using a chain of processors, the 166A may be placed either before or after effects or dynamic processors. However, if you are using the 166A for speaker protection, the compressor should be as close to the amplifier as possible in the signal chain. We recommend you use common sense and experiment with different setups to see which one provides the best results for your needs.

Note: Never connect the 166A's input to the speaker output of an instrument or power amplifier.

4. Power On the unit: Securely connect the AC power cord to the unit and mains power.

Note: Check the line voltage. The unit is shipped for 115V or 230V, 50Hz or 60Hz operation. Refer to the unit's rear panel to verify your unit's precise line voltage.

Using The SIDECHAIN INSERT

The SIDECHAIN INSERT can be used to control the compressor or the expander/gate by signals other than the audio input (via an auxiliary device, such as an equalizer). Common Sidechain applications include keyed gating, frequency-sensitive gating and frequency-weighted compression. These topics are covered in detail in the previous pages of this manual. Certain Sidechain applications may require special cabling.

For example, to set up your 166A for frequency-sensitive gating or frequency-weighted compression, you must feed an equalizer's input with the same signal fed to the 166A's INPUT, and then connect the equalizer's output to the 166A's SIDECHAIN INSERT jack (Figure 3). Providing the signal to both the 166A's INPUT and the equalizer can be accomplished in several different ways: (1) use an insert cable plugged into the SIDECHAIN INSERT jack. The SEND goes to the input of the EQ. The RETURN goes to the output of the EQ; (2) use a Y-cable to feed the audio source to both the 166A INPUT and the equalizer input; (3) feed the signal to one of the 166A's INPUT jacks and use the compressor channel's parallel INPUT jack to drive the equalizer (e.g., if the audio source feeds into Channel 1's 1/4" INPUT jack, use Channel 1's XLR INPUT jack to feed the signal to the equalizer); (4) if the audio source can internally split its output signal (e.g., some synthesizers can send the same signal from two outputs), plug a cable into each output and feed one cable to the 166A INPUT and the other to the equalizer.

Specific System Connections

The 166A has balanced inputs and outputs, and can be used with any line-level device. Some common examples include: mixing consoles, musical instruments, patch bays, and other signal processors.

Mixing Board

If you wish to compress a particular track of a multitrack recording or one channel of a live performance, connect the 166A INPUT to the audio source's output jack while the 166A OUTPUT can be directly connected to a line input jack (balanced or not) or the 166A's INPUT and OUTPUT can be wired to an Insert point. In the latter case, the signals will most likely be unbalanced.

Note: The amount of compression is directly related to the level of the input signal. However, depending upon your system's setup, it may not always be clear as to what volume controls in your chain affect input level and which affect output level. If the 166A is connected so that compression occurs before the mixer's volume controls (e.g., the 166A is connected directly between an audio source and the mixer input, or the 166A is connected to mixer inserts that are "pre-fader"), you can boost or cut the input level by adjusting the source's volume control (e.g., a synthesizer's volume control) and boost the track's output level using the 166A's OUTPUT GAIN control or the mixer's volume fader (the latter here is great for track fade-outs). However, if the 166A is connected to "post fader" mixer inserts, adjusting the mixer's volume fader changes the input level and the amount of compression. If you would rather have this volume fader control output, we suggest that you set up the compressor directly between the source and the mixer channel's input. This way, you can use the instrument's volume control to define the input level and amount of compression and the mixer's volume fader to change only the overall volume of the track.

Musical Instruments (e.g., Electric Guitar, Bass, Keyboards, Electric-Acoustic Instruments)

The output of an electric guitar is sometimes not "hot" enough to drive the 166A's INPUT. When this is the case, you should use the "PREAMP OUT" of your guitar amp (if so equipped), or the output of some other device that is designed to accept low-level instrument inputs (including various foot pedal effects, acoustic pickup preamps, and rack mount audio products, like the dbx 163X, 263X, and 563X). Such sources can be balanced or unbalanced - this is no problem for the 166A.

Microphones, bass guitars, and electric-acoustic instruments, also typically have low-level outputs. With most setups they require signal boost to drive the 166A's INPUT. For example, when recording voice directly to a portable tape deck, a mic preamp (like the dbx 760X) placed between the mic and the 166A (which is then fed to one of the recorder's inputs) can boost the signal for the 166A as well as provide a high level signal to the tape deck.

Keyboards, samplers, drum machines and sound modules typically produce a line-level signal and can be connected directly from the instrument's output to the 166A's INPUT.

Note: DO NOT CONNECT the 166A's input to the speaker output of an instrument or power amplifier. Severe damage to system components may result.

Patch Bay

In the studio, the 166A may be connected to a patch bay (such as a dbx PB-48) to allow it to be used anywhere in the studio system. If your studio is not fully balanced, you must ground the unused balanced output conductor: XLR pin (either pin 2 or 3) or the ring of a 1/4" stereo phone jack.

Sound Reinforcement

To compress a live mix or to protect loudspeakers, connect the 166A between the source (mixing board or distribution amp) and the power amp(s). If multi-way loudspeakers with low-level electronic crossovers are used, the 166A(s) should go after the crossover(s). For a stereo system, you can separately stereo couple the two high band crossovers, low band crossovers, etc. If limitations require that you use a single 166A channel before a crossover, adding an equalizer to the side chain may provide some additional protection to your high frequency components (see "Speaker Protection," page 9).

INSTALLATION CONSIDERATIONS

Input/Output Cable Configurations

Hookups and Cabling

The 166A is a balanced (differential) unit designed for nominal +4dBu levels; inputs and outputs are tip/ring/sleeve phone jacks and XLR-type jacks. The 166A can be used with either balanced or unbalanced sources and outputs can be used with either balanced or unbalanced loads, provided you use proper cabling.

A balanced line is defined as two-conductor shielded cable with the two center conductors carrying the same signal but of opposite polarity with respect to ground. An unbalanced line is generally a single-conductor shielded cable with the center conductor carrying the signal and the shield at ground potential.

Note: A given pair of channel output or input jacks (e.g., Channel 1 XLR OUTPUT and Channel 1 1/4"OUTPUT) can respectively support or drive the same signal at the same time, but if one of the jacks is unbalanced, then the other jack will be unbalanced. For example, if a 1/4" OUTPUT jack is used with a mono cable, and is therefore unbalanced, the XLR OUTPUT jack will also be unbalanced (Pin 3 shorted to ground).

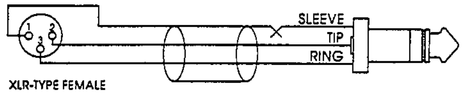
Input Cable Configurations

The 166A has an actual input impedance of >40kΩ in balanced or unbalanced configurations. This makes the 166A audio input suitable for use with virtually any source impedance, low or high. The 166A's input jacks are wired in parallel. The phone jack TIP (+) connection is internally wired to the XLR pin 2, the RING (-) is wired to pin 3, and the SLEEVE (shield) is wired to pin 1. Note that pins 2 and 3 are the reverse of certain older dbx and other manufacturer's equipment, but if the same connection is used at both the input and the output, the signal will be correctly polarized ("in phase").

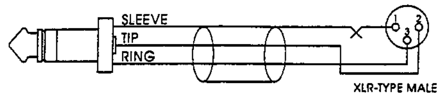
Reversing the input wires to the input terminals will result in the output signal polarity being the opposite of the input signal ("180° out of phase").

Output Cable Considerations

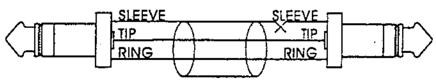
The model 166A's outputs are wired in parallel: both the XLR-type OUTPUT jack and the 1/4" stereo phone jack are capable of driving a 600Ω load. The phone jack TIP (+) connection is internally wired to the XLR pin 2, the RING (-) is wired to pin 3, and the SLEEVE (shield) is wired to pin 1. Note that pins 2 and 3 are the reverse of certain older dbx and other manufacturer's equipment, but if the same connection is used at both the input and the output, the signal will be correctly polarized ("in phase").



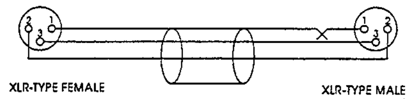
FEMALE XLR-TYPE TO STEREO PHONE PLUG



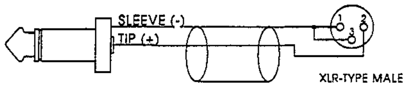
STEREO PHONE PLUG TO MALE XLR-TYPE



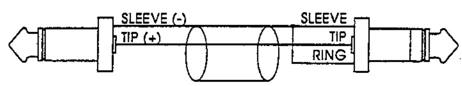
STEREO PHONE PLUG TO STEREO PHONE PLUG



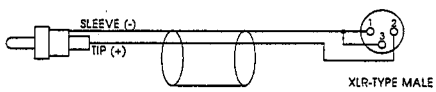
FEMALE XLR-TYPE TO MALE XLR-TYPE



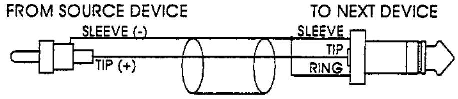
MONO PHONE PLUG TO MALE XLR-TYPE



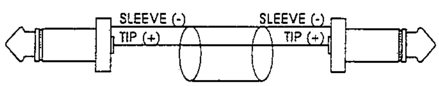
MONO PHONE PLUG TO STEREO PHONE PLUG



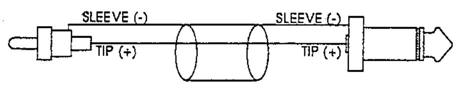
RCA PHONO PLUG TO MALE XLR-TYPE



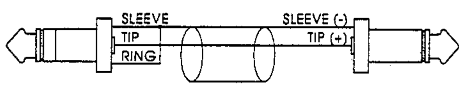
PHONO PLUG TO STEREO PHONE PLUG



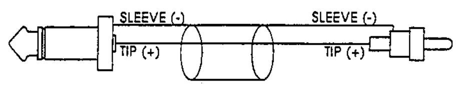
MONO PHONE PLUG TO MONO PHONE PLUG



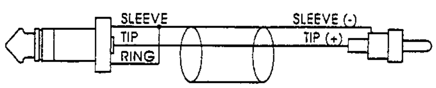
PHONO PLUG TO MONO PHONE PLUG



STEREO PHONE PLUG TO MONO PHONE PLUG



MONO PHONE PLUG TO PHONO PLUG



PHONO PLUG TO MALE XLR-TYPE

Grounding

For maximum hum rejection with a balanced source, avoid common grounding at the 166A's input and output. Most balanced (3-conductor) cables have the shield connected at both ends. This can result in ground loops which cause hum. If hum is a problem, try disconnecting the shield on one or more of your cables, preferably at the input of a device, not at the output: Ground the shield of the input cable at the source device (leaving it unconnected at the 166A's INPUT) and ground the shield of the output cable to the ground terminal of the 166A (leaving it unconnected at the receiving device). The shield is pin 1 on the XLR, SLEEVE on a 1/4" TRS.

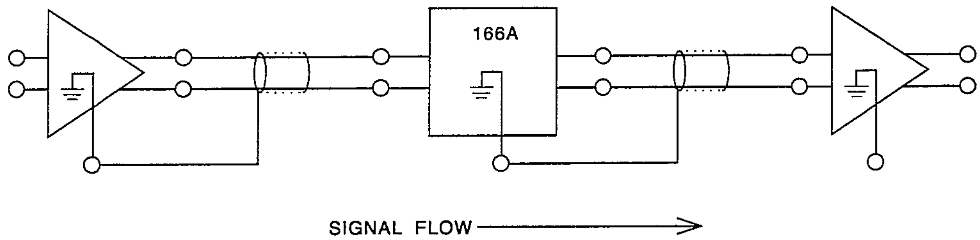


Figure 4: Signal Flow (Balanced Connection)

Technical Support, Factory Service

Technical Support/Factory Service

The 166A is an all-solid-state product with components chosen for high performance and excellent reliability. Each 166A is designed, assembled, tested, burned in and calibrated at the factory in the USA and should require no internal adjustment of any type throughout the life of the unit. We recommend that your 166A be returned to the factory only after referring to the manual and consulting with Customer Service.

Our phone number, fax number and address are listed on the inside front cover. When you contact dbx Customer Service, be prepared to accurately describe the problem. Know the serial number of your unit - this is printed on a sticker attached to the rear panel.

Note: Please refer to the terms of your Limited Two-Year Standard Warranty, which extends to the first end-user. After the warranty expires, a reasonable charge will be made for parts, labor, and packing if you choose to use the factory service facility. In all cases, you are responsible for transportation charges to the factory. dbx will pay return shipping if the unit is still under warranty.

Shipping Instructions: Use the original packing material if it is available. Mark the package with the name of the shipper, and with the following words in quotes in red: "DELICATE INSTRUMENT, FRAGILE!" Insure the package properly. Ship prepaid, not collect. Do not ship parcel post. (If you do not plan to save the packaging material, please recycle it.)

Registration Card and User Feedback

We appreciate your feedback. After you have an opportunity to use your new 166A, please complete the Registration Card and return it.

SPECIFICATIONS

Inputs (1/4" TRS Phone and XLR)	Floating Balanced; XLR: Pin 2 and TIP HI
Impedance	>40kΩ
Maximum Level	+24dBu, Balanced or Unbalanced
CMRR	>40dB
Sidechain Insert (1/4" TRS Phone)	Normalled: Ring = Send; Tip = Return
Impedance	Tip = >10kΩ (Input) Ring = 2kΩ (Output)
Maximum Level	+24dBu
Outputs (1/4" TRS Phone and XLR)	Floating Balanced; XLR: Pin 2 and TIP HI
Impedance	100Ω
Maximum Level	+20dBu, Balanced or Unbalanced
Frequency Response	20Hz - 20kHz; +0, -0.5dB,Typical 3dB points are 0.35Hz and 110kHz
Noise	<-90dBu, 22Hz to 22kHz, no weighting
THD+ N	Typically <0.04%; Any Amount of Compression Up to 40dB@1kHz
SMPTE IMD	Typically <0.08% @ +10dBu (SLOW Compression,15dB Gain Reduction)
Compressor	
Threshold Range	-40dBu to +20dBu
Threshold Characteristic	Selectable OverEasy or Hard Knee
Compression Ratio	Variable; 1:1 to Infinity:1; 50dB Maximum Compression
Attack Time	Program-Dependent; Typically 5ms (Fast MODE), 15ms (SLOW MODE) for 15dB Gain Reduction
Release Time	Program-Dependent; Typically 50dB/Sec (Fast MODE), 8dB/Sec (SLOW MODE)
Expander/Gate	
Threshold Range	OFF to +15dBu
Expansion Ratio	10:1
Maximum Depth	>50dB
Attack Time	<500μs (from Maximum Depth)
Release Time	Adjustable, 30mS to 3Sec (to 30dB attenuation)
PeakStop Limiter	
Threshold Range	0dBu to +20dBu,
Gain Adjustment Range	Variable; -20dB to +20dB
Interchannel Crosstalk	<-80dB, 20Hz to 20kHz
Dynamic Range	110dB
Stereo Coupling	True Power Summing
Operating Voltage	90-130VAC, 50/60Hz; 200-250VAC,50/60Hz
Operating Temperature	0°C to 45°C (32°F to 113°F)
Dimensions (H x W x D)	1.75" x 19" x 6.5" (4.45cm x 48.2cm x 16.51cm)
Rack Space	1 Rack Unit (1U High)
Weight	Net Weight: 4.85lb (2.19 kg) Shipping Weight: 7.10lb (3.21 kg)

Notes: Noise and frequency response specifications are at unity gain.
0dBV =1.0Vrms; 0dBu =0.775Vrms.
Specifications are subject to change.