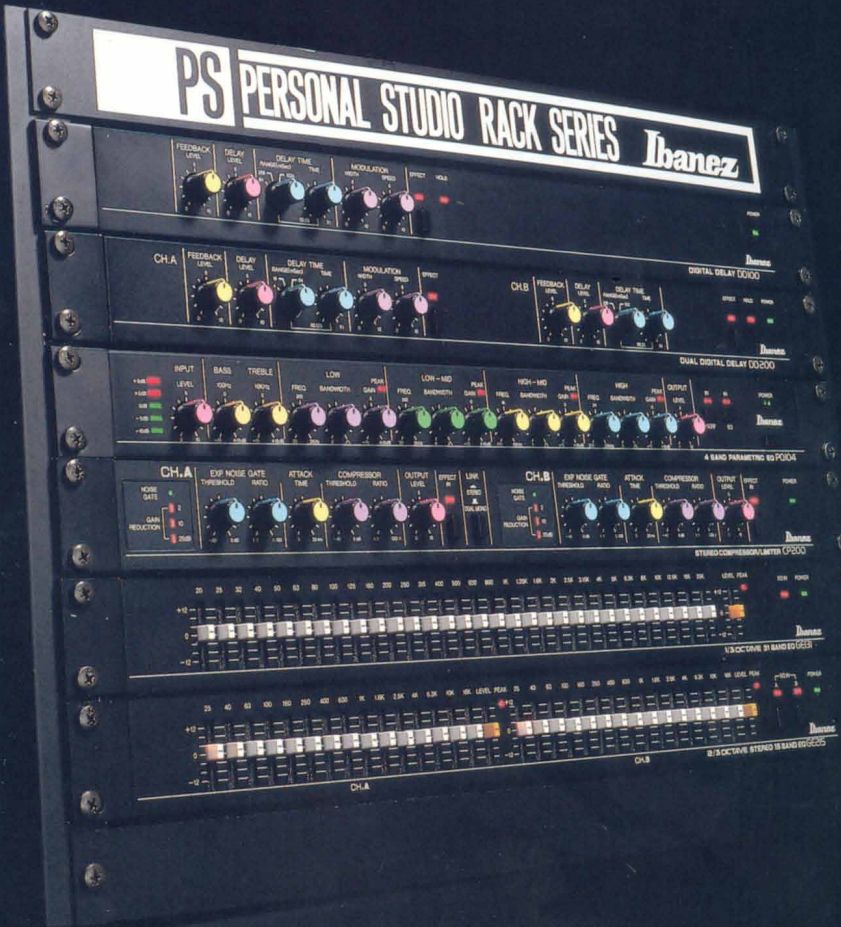


PS PERSONAL STUDIO RACK SERIES *Ibanez*

HAND BOOK



Iba

PERSONAL S

S E R

INTRODUCTION

Welcome to the exciting world of the Personal Studio Rack Series. This line of signal processors includes two digital delays, three equalizers, and a compressor-limiter; all of single-space (1 3/4" high) standard rack mount design (19" wide). Heavy-duty but light-weight metal housings are extremely compact, saving rack space in depth as well as height.

These units are technically adaptable to a wide variety of uses including all types of personal recording systems, instrument amplification racks and many sound reinforcement applications. Level switches provide -10dBv or -20dBv input/output choices on all units except the PQ104, which has variable input and output level controls. All units have both 1/4 phone and RCA phone jacks available at a variety of patch points for standard and creative applications. Remember that instruments and microphones need to be pre-amped before being processed to keep noise to a minimum. Use a mixer or instrument pre-amp with this series for optimum results.

Specifications and quality of the Personal Studio Series rivals professional units costing considerably more. These models make it possible for musicians to add the kind of quality and creativity to their music that could only be achieved by top engineers not too long ago. External power supplies (multi-unit chaining) in addition to the most current technology, efficient design and manufacture, are responsible for the low noise, full bandwidth and dynamic range of this series. Whether your application is live or recording and no matter what instrumentation is being used, the Personal Studio Rack Series will provide you with professional signal processing to meet your needs.



mezz

STUDIO RACK

UNIVERSALS

● DD100 DIGITAL DELAY



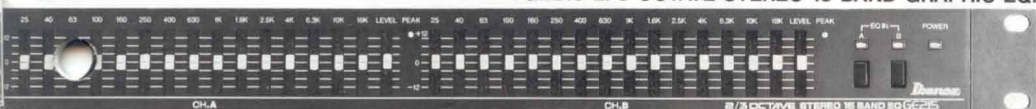
● DD200 DUAL DIGITAL DELAY



● GE131 1/3 OCTAVE 31 BAND GRAPHIC EQ.



● GE215 2/3 OCTAVE STEREO 15 BAND GRAPHIC EQ.



● CP200 STEREO COMPRESSOR / LIMITER



● PQ104 4 BAND PARAMETRIC EQ.



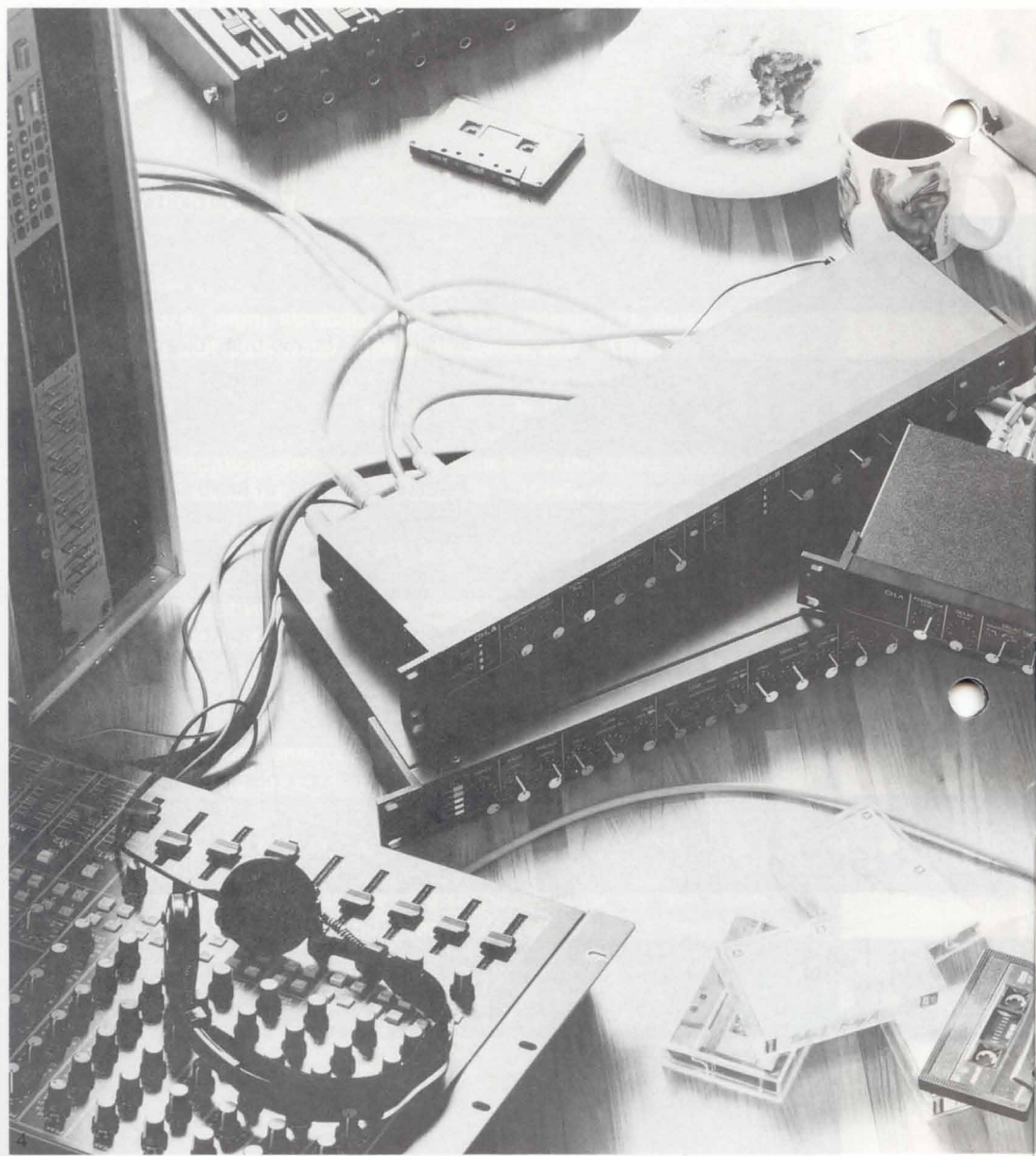


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1 DD100/DD200

DIGITAL DELAY / DUAL DIGITAL DELAY

■ FEATURES AND PRINCIPLES

The DD100 and DD200 digital delays provide high quality time based effects such as flanging, chorus, doubling, slapback, short and long delays, plus hold. The DD100 is of the standard format found on most digital delays currently in use. It gives control over a delay range from 16 to 4096 ms. (more than 4 seconds). This is considerably longer than most models and is especially useful in the hold mode.

A modulation section allows automatic changing of delay time by an L.F.O. (Low Frequency Oscillator). Changing speed on a tape recorder affects the pitch of a sound. Digital delays are recorders that automatically play back after a certain amount of time set by the delay time controls. Modulating the delay time causes pitch to go sharp and flat of the original signal. By setting a starting point of approximately 30ms., which is not perceived as an echo since the two sounds seem to occur simultaneously, then adding modulation, pitch change effects are available. The width control sets the range of pitch fluctuation while the speed control adjusts the rate at which the cycle repeats. This signal along with the dry signal can create a range of flanging and chorus effects that "add life" to the sound. Feedback allows the "feeding back" of the delayed signal back into the circuit so it can be delayed again. The higher the setting the more repeats. On short modulated delay times this adds emphasis to the effect. On longer delay times the feedback actually increases the number of repeats after the original signal. With the proper footswitch plugged into the hold input the footswitch depression will cause the sound to playback or repeat indefinitely until switched off. Delay time should be set medium to high for this effect.

A variety of outputs allows different types of stereo

effects. Using the dry and mix outputs only one side has the effected signal. This is popular for echo and hold effects. On chorus settings, some studio musicians also use this type of stereo when recording master sessions because it keeps phasing side-effects to a minimum. Using the Mix and Invert Mix outputs creates a more animated chorus or flanging effect by introducing a 180 degree phase shift on one side. On flanging settings the invert mix output will have slightly more resonance than the mix out.

The DD200 actually is two independent digital delay units in one chassis. Channel A has a modulation section, and a maximum delay time of 256 ms. This channel can be used for a variety of effects, including: flanging, doubling, chorus, vibrato, or short delays. Channel B has a maximum delay time of 2048 ms., and is most suited for medium to long delay effects. This channel, along with the footswitchable "Bypass", features a versatile "Hold" function also.

By using the output patching section creatively, quite a variety of applications exist, including:

1. Using the two units on separate instruments with totally different settings.
2. Patching (in series) one channel into the other for 2 effects simultaneously.
3. Patching (in series) one channel into the other for 2 different presets footswitchable.
4. Stereo effects by having different settings on the left and right sides of a mix.

Dozens of combinations are possible in patching (Dry A into B with Mix A on Left while Mix B is on Right, Dry A to left while Mix A goes to B then Mix B to the left while the right side is comprised of invert mix from A, etc.). Use your imagination (and a mixer).

● DD100 DIGITAL DELAY (FRONT/REAR)



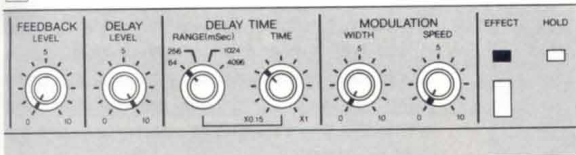
● DD200 DIGITAL DELAY (FRONT/REAR)



USE

DD100

1 SUBTLE STEREO EFFECT



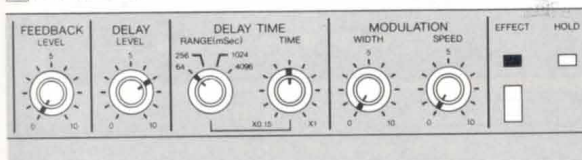
In example 1 a subtle stereo effect can be added to any sound giving depth especially effective on background vocals, heavy metal rhythm guitar and lead solos on any instrument.

2 CHORUS



The chorus effect in example 2 will sweeten up any type of guitar or electric piano sound. Synthesizer programs for strings and organ are drastically improved in realism with this setting. Experiment with settings as noted to suit your instrument and taste.

3 DOUBLING



Example 3 for doubling gives the impression if used in stereo of two identical instruments and parts being played simultaneously. This is helpful for solos, backup vocals, handclaps, choir and brass sounds.

4 SLAP-BACK



Slapback in example 4 was used in many early rock and roll recordings on the lead vocal and continues today to be very popular. To simulate the old tape echo units put an equalizer on the delayed part of the sound and subtract some high frequencies.

5 ECHO



Example 5 gives the general setting areas for a wide variety of echo and delay effects. Experiment to suit your needs, keeping song tempo in mind and trying to incorporate the delay time into the musical context.

6 HOLD

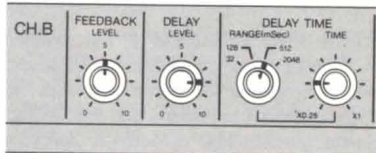
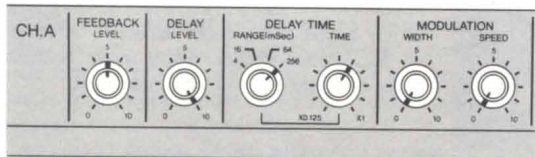


Context is extremely important in example 6 as the delay time and depression of the hold pedal must be precisely synchronized with the end and re-start of the musical passage being played. After the hold is successfully "looped" you can play along with the passage. An interesting application is to use another tone on the instrument giving the impression of two musicians playing. After mastering use of the hold function you may find that some of the same techniques can be used with regular echo settings by playing "to the echo" and planning harmonies that follow one another.

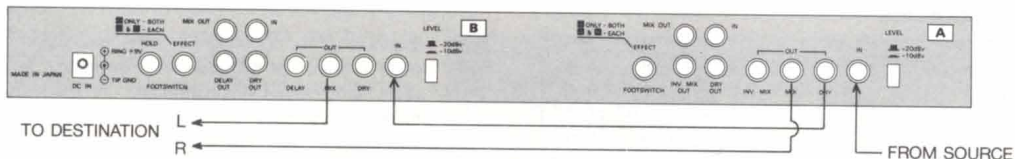
USE

DD200

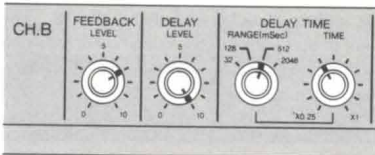
1 STEREO ECHO



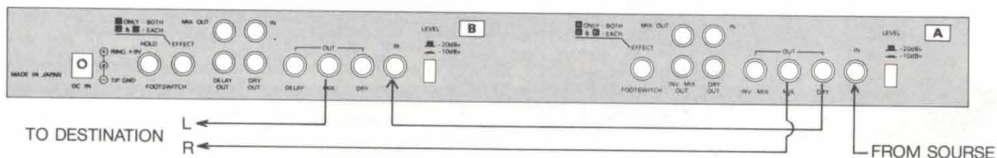
Example 1 shows settings for a stereo echo effect which has more life to it than just a single unit echo because, although the delay times on both units are approximately the same, the slight difference seems to widen the field of sound.



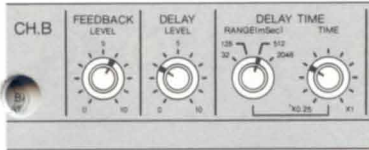
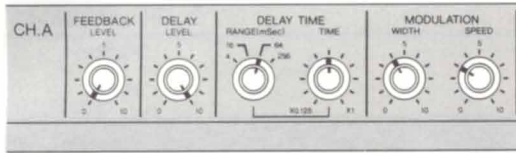
2 MULTI-TAP DELAY



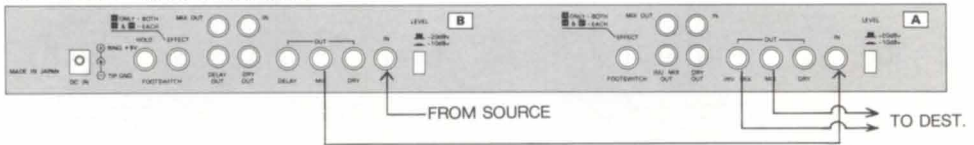
Multi-tap delay as in example 2 is similar except there is more difference between the delay times and the added feedback, creates the perception of a random pattern of echos.



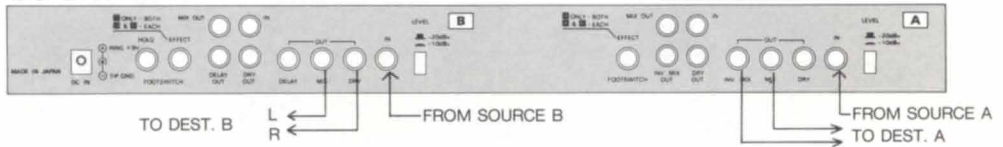
3 CHORUS & DELAY



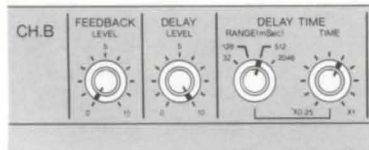
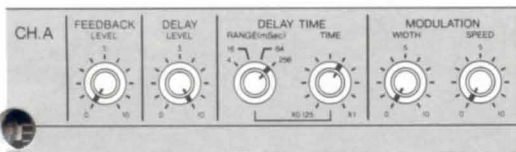
BOTH EFFECTS ON ONE INSTRUMENT



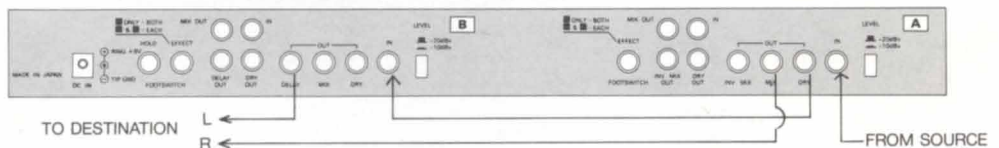
EACH EFFECT USED ON DIFFERENT INSTRUMENT



4 STEREO PING-PONG



The stereo ping-pong effect in example 4 is achieved by setting the delay time on Channel A to exactly half the time of Channel B. The effect is like having the Channel A delay time setting with 2 repeats; one on the left followed by one on the right. Adding feedback to either A or B can cause lingering effect in which the stereo is less obvious than the first 2 repeats.





















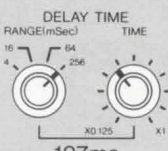







■ DELAY TIME CHART

DD100

TABLE 3 (CONTINUED)

NOTE				
TEMPO				
= 80	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>750ms</p>	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>375ms</p>	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>187ms</p>	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>250ms</p>
= 100	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>600ms</p>	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>300ms</p>	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>150ms</p>	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>200ms</p>
= 120	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>500ms</p>	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>250ms</p>	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>125ms</p>	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>166ms</p>
= 130	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>462ms</p>	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>231ms</p>	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>115ms</p>	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>154ms</p>
= 140	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>429ms</p>	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>214ms</p>	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>115ms</p>	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>143ms</p>
= 155	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>400ms</p>	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>200ms</p>	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>100ms</p>	<p>DELAY TIME</p> <p>RANGE(mSec) TIME</p> <p>133ms</p>

NOTE TEMPO				
 = 80	————	————	 187ms	 250ms
 = 100	————	————	 150ms	 200ms
 = 120	————	 250ms	 125ms	 166ms
 = 130	————	 231ms	 115ms	 154ms
 = 140	————	 214ms	 107ms	 143ms
 = 155	————	 200ms	 100ms	 133ms

DD200 CH. B

NOTE TEMPO				
$\text{♩} = 80$	<p>DELAY TIME RANGE(mSec) TIME</p> <p>750ms</p>	<p>DELAY TIME RANGE(mSec) TIME</p> <p>375ms</p>	<p>DELAY TIME RANGE(mSec) TIME</p> <p>187ms</p>	<p>DELAY TIME RANGE(mSec) TIME</p> <p>250ms</p>
$\text{♩} = 100$	<p>DELAY TIME RANGE(mSec) TIME</p> <p>600ms</p>	<p>DELAY TIME RANGE(mSec) TIME</p> <p>300ms</p>	<p>DELAY TIME RANGE(mSec) TIME</p> <p>150ms</p>	<p>DELAY TIME RANGE(mSec) TIME</p> <p>200ms</p>
$\text{♩} = 120$	<p>DELAY TIME RANGE(mSec) TIME</p> <p>500ms</p>	<p>DELAY TIME RANGE(mSec) TIME</p> <p>250ms</p>	<p>DELAY TIME RANGE(mSec) TIME</p> <p>125ms</p>	<p>DELAY TIME RANGE(mSec) TIME</p> <p>166ms</p>
$\text{♩} = 130$	<p>DELAY TIME RANGE(mSec) TIME</p> <p>462ms</p>	<p>DELAY TIME RANGE(mSec) TIME</p> <p>231ms</p>	<p>DELAY TIME RANGE(mSec) TIME</p> <p>115ms</p>	<p>DELAY TIME RANGE(mSec) TIME</p> <p>154ms</p>
$\text{♩} = 140$	<p>DELAY TIME RANGE(mSec) TIME</p> <p>429ms</p>	<p>DELAY TIME RANGE(mSec) TIME</p> <p>214ms</p>	<p>DELAY TIME RANGE(mSec) TIME</p> <p>107ms</p>	<p>DELAY TIME RANGE(mSec) TIME</p> <p>143ms</p>
$\text{♩} = 155$	<p>DELAY TIME RANGE(mSec) TIME</p> <p>400ms</p>	<p>DELAY TIME RANGE(mSec) TIME</p> <p>200ms</p>	<p>DELAY TIME RANGE(mSec) TIME</p> <p>100ms</p>	<p>DELAY TIME RANGE(mSec) TIME</p> <p>133ms</p>

■ SPECIFICATIONS

DD100

● DELAY TIME	1: 10 msec - 64 msec
	2: 43 msec - 256 msec
	3: 170 msec - 1024 msec
	4: 683 msec - 4096 msec
● MODULATION SWEEP SPEED	0.06 Hz - 13 Hz
● MODULATION SWEEP RATIO	1:6
● FREQUENCY RESPONSE	20 Hz - 12 kHz (+0, -3 dB)
● TOTAL HARMONIC DISTORTION	0.3%(INPUT 400 Hz, -20 dB)
● EQUIVALENT INPUT NOISE	-100 dBv(IHF-A INPUT SHORTED)
● INPUT IMPEDANCE	1 Mohms(-20 dBv)/(-10 dBv)
● OUTPUT IMPEDANCE	1 kohms(-20 dBv)/5 kohms(-10 dBv)
● CURRENT CONSUMPTION	190mA(DC 9V)
● DIMENSIONS	482(W) × 141(D) × 44(H)mm
● WEIGHT	1.4 kg

DD200

CHANNEL A	
● DELAY TIME	1: 0.5 msec - 4 msec
	2: 2 msec - 16 msec
	3: 8 msec - 64 msec
	4: 32 msec - 256 msec
● MODULATION SWEEP SPEED	0.06 Hz - 13 Hz
● MODULATION SWEEP RATIO	1:8
CHANNEL B	
● DELAY TIME (HOLD TIME)	1: 8 msec - 32 msec
	2: 32 msec - 128 msec
	3: 128 msec - 512 msec
	4: 512 msec - 2048 msec
● FREQUENCY RESPONSE	20 Hz - 12 kHz (+0, -3 dB)
● TOTAL HARMONIC DISTORTION	0.3%(INPUT 400 Hz, -20 dB)
● EQUIVALENT INPUT NOISE	-100 dBv(IHF-A INPUT SHORTED)
● INPUT IMPEDANCE	1 Mohms(-20 dBv)/47 kohms(-10 dBv)
● OUTPUT IMPEDANCE	1 kohms(-20 dBv)/5 kohms(-10 dBv)
● CURRENT CONSUMPTION	190mA(DC 9V)
● DIMENSIONS	482(W) × 141(D) × 44(H)mm
● WEIGHT	1.6 kg

2 GE131/GE215

1/3 OCTAVE 31 BAND EQ. / 2/3 OCTAVE STEREO 15 BAND EQ.

■ FEATURES AND PRINCIPLES

The GE131 and GE215 equalizers allow the user to see a "graphic" representation of the frequency response curve which he can impose on incoming audio signals. Reading left to right starting with the lowest frequencies a gain control slider is provided for each center frequency. The number above each slider represents the frequency at the center of the band affected. The width of each band is 1/3 of an octave on the GE131 and 2/3 of an octave on the GE215, which when applied to the entire audio spectrum of 20-20KHZ, divides into 31 and 15 bands, respectively. Each gain control can add or subtract up to 12dB (decibels) of volume (amplitude) in that frequency range. A master level control is also provided to counteract a change in overall volume when using extreme equalization settings. All gain sliders are notched for easy reference to the flat or zero change position. The GE215 actually has two independent but identical equalizers in one unit for stereo or dual processing. Peak indicator LEDs on both models provide quick warning that incoming

levels are too high and an EQ in/out switch allows quick comparison to the original signal.

Graphic equalization is useful for both tone shaping and control of feedback or problem areas. When a stereo mix that is not very prone to feedback needs to be "shaped", such as a house P.A. system or mix-down of a multitrack tape, the GE215 is the perfect tool. The GE131 will be more useful in a monitor system because of its precise control of so many different bands in the audio spectrum. Two GE131 units will be required in stereo systems. Sound reinforcement monitor systems are feedback prone because of the high levels needed for vocals to compete with acoustic drums and loud instrument amplification. Precise control of subtractive equalization at the problem frequency is necessary. In studio monitoring applications, equalization can help attain a flat playback system when the speaker or room isn't reproducing or reflecting an even response throughout the audio spectrum.

● GE131 1/3 OCTAVE 31 BAND EQ. (FRONT/REAR)



● GE215 2/3 OCTAVE STEREO 15 BAND EQ. (FRONT/REAR)



USE

GE215

1 STEREO SOUND SHAPING

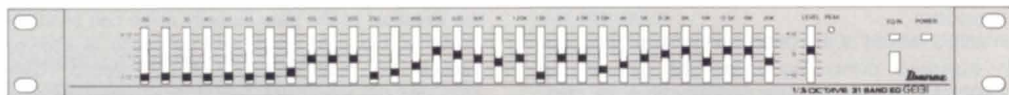


In example 1 the GE215 has been used to generally shape the tone of a stereo sound system. Gradual reduction of low frequencies helped to "clean up" a "fuzzy" sounding room. The drastic reduction at 250 Hz. was necessary because of a feedback in that range. At 1.6K another slight cut helped smooth out a harsh sounding metal mid-range horn. As the crowd moved

in it was necessary to boost high frequencies to overcome the absorption in that area, however no boost was applied at 16K since the roll-off of the high frequency driver was much lower than 16K. Adding here might cause distortion as the driver could heat up attempting to reproduce these frequencies.

GE131

2 FEEDBACK CONTROL



The GE131 used in example 2 helped to overcome a major feedback problem in a stage monitor that was very close to both a vocal microphone and a brick wall. Since the speaker had a 12" woofer and only vocals were in this monitor mix, all lows below 125 Hz were removed since they were unnecessary. A low frequency rumble still existed so, after trial and error,

it was decided to have been around 200 – 300 Hz and those nearest bands were subtracted. Other feedback areas were at 1.6KHz and in the 3 – 5KHz region. 10KHz was subtracted to keep distortion down in the tweeter while 20KHz was subtracted because the tweeter rolled off at 18KHz. Overall level was increased to counteract all the subtractive e.g.

GE215

3 DUAL PROCESSING (Kick and Snare)



In example 3 channel one of the GE215 is used for kick drum while channel two is for snare. Lows boosted on both channels give a fullness to each drum in it's proper frequency range. The boost in the mids on

channel one gives more attack and cutting power to the kick. Subtracted mids on the snare channel removes the "boxy" sound and emphasizes the crisp tonality created by boosting in the high frequency range.

SPECIFICATIONS

GE131

- FREQUENCY RESPONSE 20 Hz – 20 kHz (±1 dB) @ –30 dBv
(ALL EQUALIZATION CONTROLS AT FLAT)
- EQUIVALENT INPUT NOISE –105 dBv (–20 dBv) / –95 dBv (–10 dBv) HF-A
(ALL EQUALIZATION CONTROLS AT FLAT)
- MAXIMUM INPUT LEVEL +9 dBv @ 1 kHz THD = 3%
- INPUT IMPEDANCE 1 Mohms (–20 dBv) / 47 kohms (–10 dBv)
- OUTPUT IMPEDANCE LESS THAN 10 kohms
- EQUALIZATION CONTROL (1/3 OCTAVE)
..... CENTER FREQUENCY ±7% (MAX)
20, 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200, 250,
315, 400, 500, 630, 800, 1k, 1.25k, 1.6k, 2k, 2.5k,
3.15k, 4k, 5k, 6.3k, 8k, 10k, 12.5k, 16k, 20k
RANGE OF BOOST/CUT ±12 dB
- PEAK INDICATOR
..... TURN ON WHEN THE OUTPUT LEVEL REACHES +4 dBv
- CURRENT CONSUMPTION 70mA (DC 9V)
- DIMENSION 482(W) × 132(D) × 44(H)mm
- WEIGHT 1.4 kg

GE215

- FREQUENCY RESPONSE 20 Hz – 20 kHz (±1 dB)
@ –30 dBv (ALL EQUALIZATION
CONTROLS AT FLAT)
- EQUIVALENT INPUT NOISE –105 dBv (–20 dBv) / –95 dBv (–10 dBv) HF-A
(ALL EQUALIZATION CONTROLS AT FLAT)
- MAXIMUM INPUT LEVEL +9 dBv @ 1 kHz THD = 3%
- INPUT IMPEDANCE 1 Mohms (–20 dBv) / 47 kohms (–10 dBv)
- OUTPUT IMPEDANCE LESS THAN 10 kohms
- EQUALIZATION CONTROL CENTER FREQUENCY ±7% (MAX)
(2/3 OCTAVE)
25, 40, 63, 100, 160, 250, 400, 633, 1k, 1.5k, 2.5k, 4k,
6.3k, 10k, 16k
RANGE OF BOOST/CUT ±12 dB
- PEAK INDICATOR TRUN ON WHEN THE OUTPUT LEVEL REACHES
+4 dBv
- CURRENT CONSUMPTION 80mA (DC 9V)
- DIMENSIONS 482(W) × 132(D) × 44(H)mm
- WEIGHT 1.4 kg

3 CP200

STEREO COMPRESSOR / LIMITER

FEATURES AND PRINCIPLE

Like other compressor-limiters the CP200 is able to compress the dynamic range of an audio signal and to limit transients from rising above a certain level. Common applications are:

1. To keep from overloading an input of a mixer or tape recorder when using instruments that have extremely wide dynamic range or attack transients.
2. To keep levels consistent when signal is fluctuating too much as in bass guitars with live and dead spots.
3. To simulate sustain in an instrument whose decay is too short for a certain musical context, such as lead guitar.

There are two independent channels in the CP200 which can operate on different signals in different ways simultaneously or can be linked in stereo so transients on either side will trigger the effect of both channels, keeping both sides even throughout the program.

An added feature of the CP200 is the noise gate. As on expensive compressor-limiters the gate of the CP200 has its own controls separate from the compressor section. LED indicators for each section give the status of the effect.

The most important controls are attack, threshold, and ratio. Attack time sets the amount of time before full effect is reached. Set on zero, the compression starts immediately upon detection of a signal at the input. At 20ms (maximum) attack transients of the original sound are unaffected but compression quickly takes over the remainder of the sound. This is helpful on instruments such as guitar to maintain the important attack characteristic while the compression can raise the low sustaining level up for more perceived sustain. Threshold sets the minimum level needed to

start the compression cycle. At high settings (clockwise) only loud transients will be affected, whereas the lowest setting will cause compression on any signal applied. The Ratio control sets the amount of change in level in relation to the original. For example, a 2:1 ratio would cause any signal above the threshold level to increase in volume half as much as the original, thus a 6dB change at the input would be only 3dB different at the output. Hard limiting to trim transients is achieved at the highest setting.

The output control is used to maintain unity through the unit once the desired effect has been achieved. An effect bypass switch allows comparison for setting levels.

When using low threshold settings, to cause compression on low level signals, unwanted low level signals such as noise can be increased in between musical passages. Gating can solve this problem. By setting the threshold of the gate properly noise can be eliminated as the gate shuts and no sound is allowed through until more desired signal is applied. Gate ratio sets the amount of drop in level when the gate is closed. If the gate is trimming off the last part of an important signal turn this control slightly counterclockwise until this level is set slightly above the level of the desired low signal.

Release times on both the gate and the compressor are variable, and are automatically adjusted following the input level.

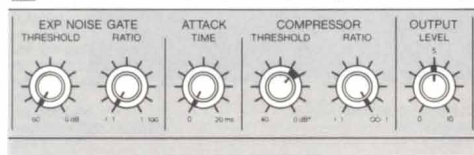
The gate can also be used as an effect itself when low level signals need to be removed from otherwise very high level programs, or to impose an abrupt decay on an instrument that normally has some sustain.

● CP200 STEREO COMPRESSOR / LIMITER (FRONT/REAR)



■ USE

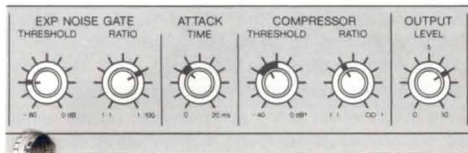
1 HAND LIMITING FOR VOCALS, SLAP BASS, DRUMS, PIANO, ETC.



Limiting, as in example 1, is useful when transient oriented material is to be recorded or amplified and the transients are overloading inputs. Especially applicable on vocals, drums, slap bass, and piano, it is also helpful when mixing or copying to in-expensive cassette decks that can't handle the dynamic range contained on the master tape. With the CP200 bypass-

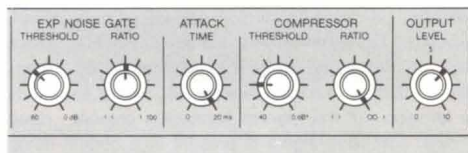
ed, and the signal overdriving an input, insert the effect with the ratio set full, attack time at zero, and threshold at zero. Watching the gain reduction display, turn the threshold counter clockwise until the LEDs light only on the loudest notes. Check input meter on mixer/recorder to verify reduction of transients. If level is still too high for input, try reducing the output level on the CP200 and/or a lower threshold setting.

2 BASS COMPRESSION



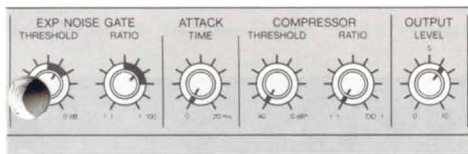
Example 2, Compression for Bass, keeps an evenness to loud and soft bass notes without losing the character of the instrument. By raising the attack control some of the initial transients are allowed through while the sustained portion of the sound is subtly compressed (4 or 5 to 1 ratio), smoothing out inconsistencies in note level. A low threshold setting allows soft and loud notes both to initiate compression. Output level may be raised to restore lost volume; however, if noise also seems louder, use the gate. A low gate threshold setting will keep the gate from closing on soft notes. Too high of a ratio on the gate might bring attention to its effect, yet a low setting might not achieve any noise reduction.

3 ADDED SUSTAIN FOR GUITAR



When using the CP200 with guitar, as in example 3, it helps to lengthen the attack time, allowing attack characteristics to go unchanged. The higher the ratio of compression the more apparent the sustain, as long as the threshold is low enough to let the decaying signal to be affected. Some gating may be necessary for noise control, however too high of a setting on the gate ratio and threshold will result in an unnatural release of the note.

4 GATING FOR SNARE, REVERB, SAMPLING, ETC.



Gating alone in example 4 is useful on a snare or drum kit to remove the snare rattles or other mechanical noises in between actual strikes of the drum and to generally "tighten" the sound of the kit.

Gating after a reverb unit can impose a non-linear decay to a heavily reverberant setting. This effect is especially useful on drum sounds when an electronic character is needed from an acoustic instrument.

This type of unnatural ambience is usually associated with digital reverbs that have a non-linear mode but can be achieved with any type of reverb and the CP200. Gates can be very helpful when using a digital sampler. Connect the CP200 between the source and the sampler's input, and noises before and after the desired sound won't be sampled along with it. This makes for cleaner samples that are easier to work with. Some compression or limiting can also help get the best sample. Loss of dynamic characteristics can be re-synthesized on most samplers via the envelope generators.

SPECIFICATIONS

- CHANNEL 2 CHANNEL
- MODE STEREO/DUAL MONO
- FREQUENCY RESPONSE 20 Hz - 20 kHz (±1 dB)
- TOTAL HARMONIC DISTORTION LESS THAN 0.05% (1 kHz, -10 dBv)
- EQUIVALENT INPUT NOISE -94 dBv (IHF-A)
- INPUT IMPEDANCE 500 kohms (-20 dBv) / 30 kohms (-10 dBv)
- OUTPUT IMPEDANCE 470 ohms
- MAXIMUM LIMITING -60 dB
- GAIN REDUCTION INDICATOR 3 POINT LED
- COMPRESSOR
 - THRESHOLD -40 dBv - 0 dBv
 - RATIO 1:1 - ∞:1
- EXPAND NOISE GATE
 - THRESHOLD -60 dBv - 0 dBv
 - RATIO 1:1 - 1:100
- THRESHOLD RESPONSE SOFT KNEE
- ATTACK TIME (10 dB) 15 msec - 35 msec
- RELEASE TIME 25 msec / dB
- CURRENT COMSUMPTION 95mA (DC 9V)
- DIMENSIONS 482(W) × 138(D) × 44(H)mm
- WEIGHT 1.5 kg

4 PQ104

4 BAND PARAMETRIC EQ.

■ FEATURES AND PRINCIPLES

The PQ104 equalizer gives the user 6 bands of control to reshape the frequency response curve of incoming audio signals. Four of these bands are totally user controllable in all 3 of the important parameters (explaining the name "parametric"). The first parameter is a frequency control which allows selection of a center frequency to affect. The Bandwidth shapes the affected area. A narrow shape allows centering in on a specific frequency where as a wide shape gives a range of frequencies up to 4.3 octaves (2 octaves on each side of the center frequency).

The amount of increase or decrease in volume (amplitude) in the selected range is set by the 3rd parameter which is the gain control. Up to 15dB of boost or cut is available. Four bands of parametric EQ give plenty of control over most audio signals but just in case you need them, 2 extra bands are provided. These controls have preset frequencies of 100Hz and 10KHz called bass and treble. Unlike the parametric, the bands are preshaped to gradually increase in amplitude as the center frequency is neared, then level off and continue the effect throughout the rest of the audio range. Thus the bass control can boost or cut low frequencies from 100Hz on down and the treble from 10KHz on up.

Since equalization can be a major source of noise in any system every possible noise reducing feature is provided. An input level control with LED indicators allows setting of the incoming signal's level to the level that is optimum for the equalizer.

Once adjustments to the equalizer are made the levels

may need to be readjusted since equalization is actually changing levels. Additional LED peak indicators show which EQ Band is being overdriven. The output level is an overall control that can be used to counteract any changes in volume caused by extreme equalization. Comparison to the original signal is quickly obtained by using the in/out switch.

Another switch called the HPF (high pass filter) provides still more equalization by passing all frequencies above 40Hz, where many sub-harmonic noises reside that are harmful to speakers and are unnecessary to musical signals.

When using the PQ104 for tone shaping, wider bandwidth settings provide a smoother sound, once the frequency range has been selected. Narrow bandwidth is useful for reducing feedback or other unwanted sounds at a certain frequency. Since the 4 bands overlap somewhat it is possible to cover the same area with 2 of the equalizer sections when extreme feedback problems occur at one frequency. When shaping a sound the lower bands boosted can add warmth and fullness, mid-range bands can add presence and clarity, added highs can increase intelligibility and put an edge on the sound while the extremely high ranges will be effective in adding "sparkle" or "sizzle" to sounds by bringing out the highest harmonics present. Keep in mind that if no signal is present in a certain range, then boosting that range increases only whatever noise resides there and will have a destructive effect rather than improvement.

● PQ104 4 BAND PARAMETRIC EQ. (FRONT/REAR)



■ USE

1 POST DISTORTION GUITAR PROCESSING



In example 1 the PQ104 is used in a chain of effects after a distortion pre-amp, but before delay and chorus. This particular setting example attempts to re-create a large tube type amplifier stack. The low boost is to simulate rumble associated with large speaker cabinets. Subtraction of low-mids further emphasizes the effect. Added mids around 800Hz. gives a singing quality to

the distortion sound. An edge is created in the high region by extreme boost with narrow bandwidth. Manually sweeping the frequency control while playing allows you to center in on the best frequency.

2 VOCAL FOR RECORDING



Example 2 shows the setting after a particular vocal is recorded. The microphone was set for recording purposes. The low band added warmth, while the low-mid band helped eliminate popping "P"s. With a fairly narrow bandwidth, the gain was cut to -15dB , then by manually sweeping the frequency control (while listening to the vocal) till the "P"s were lower in volume, the correct frequency

was found. Since other sounds were adversely affected the gain control was increased to a compromise setting. The high-mid band brought clarity and presence in while the high band eliminated some harshness. The treble tone control allowed some brilliance to be emphasized, improving intelligibility.

3 FEEDBACK CONTROL



Feedback control, as in the extreme case of example 3, is one of the most useful applications of the PQ104. A very predominant roar was so loud in this sound system that, even though the frequency was found using the low-band, it took both the low and low-mid bands to rid the system of this feedback frequency. Nearly 30dB of cut had been applied to a very narrow portion of the bass section of the frequency

spectrum. This caused a thin sound which the bass control was able to offset. Feedback was also strong in the mid-range area but no specific frequency seemed to be the culprit, so a general subtraction of mids solved this problem. A squealing high frequency was subtracted and boost added on the treble tone control to counterbalance.

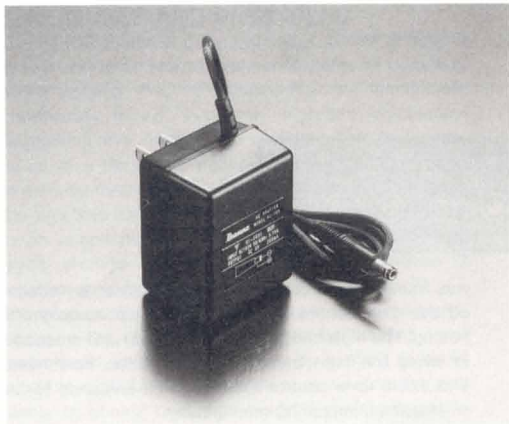
OTHERS

Other areas the PQ104 could be helpful include:

1. Adding brightness to cymbals with one band while removing a resonant overtone or subharmonic with another.
2. Finding the boominess in a miked acoustic guitar and eliminating it.
3. Getting rid of 60 cycle hum in a poorly grounded system by cutting at 60Hz.
4. Extracting the frequency of a squeaking kick drum pedal.
5. Pre-emphasis of highs when recording on a tape recorder or sampler known to rolloff high end.
6. Filtering of frequencies that cause aliasing noise in samplers.
7. Additive and subtractive assistance for creating sounds on synthesizers.

SPECIFICATIONS

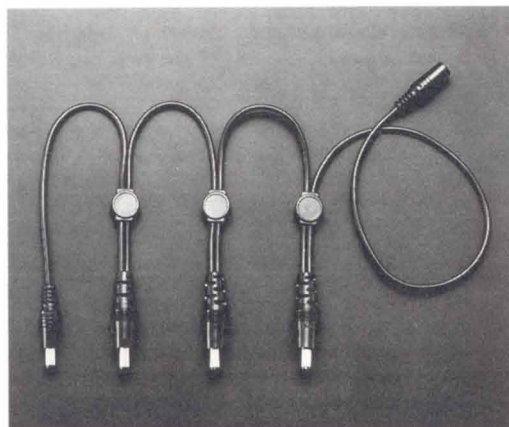
- INPUT IMPEDANCE 47 kohms
- OUTPUT IMPEDANCE LESS THAN 10 kohms
- FREQUENCY RESPONSE 20 Hz - 20 kHz (FLAT)
- EQUIVALENT INPUT NOISE -95 dBV (IHF-A, FLAT)
- CENTER FREQUENCY LOW 20 Hz - 2 kHz
LOW-MID 50 Hz - 10 kHz
HIGH-MID 100 Hz - 10 kHz
HIGH 200 Hz - 20 kHz
- BANDWIDTH 0.7 - 4.3 OCT
- GAIN LIMIT $\pm 15\text{ dB}$
- CONTROL INPUT LEVEL $\times 1$
FREQUENCY $\times 4$
BANDWIDTH $\times 4$
GAIN $\times 4$
OUTPUT LEVEL $\times 1$
TONE BASS 100 Hz $\pm 15\text{ dB}$
TREBLE 10 kHz $\pm 15\text{ dB}$
- SWITCH HIGH PASS FILTER (OCT18 dB/40 Hz)
EFFECT (NORMAL/PARAMETRIC EQ)
POWER
- JACK INPUT (PHONE, PIN)
OUTPUT (PHONE, PIN)
- INDICATOR POWER
EFFECT IN
INPUT LEVEL
HIGH PASS FILTER IN
- POWER EQUIPMENT 120V AC 60Hz 8W, 220 - 240V AC 50Hz 10W
- DIMENSION 482(W) \times 222(D) \times 44(H)mm
- WEIGHT 3.0 kg



AC109 AC ADAPTER

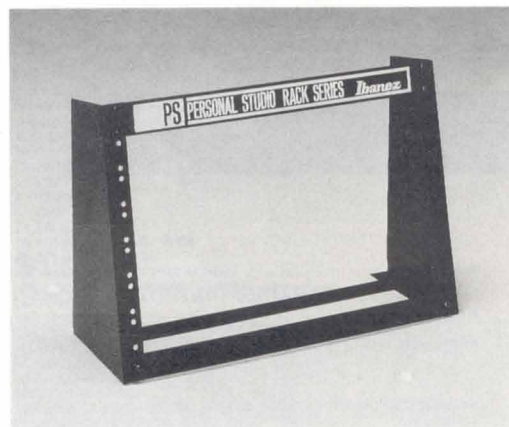
The AC109 AC adapter is the optional power supply available for "Personal Studio Rack Series" Effects. It is a 200mA regulated power supply that is suggested for extended use situations.

NOTE: The use of any AC adapter, other than the AC109 may damage or impede the performance of any "Personal Studio Rack Series" effect.



DC4 DC CORD

The DC4 operates as an extension cord to power up to 4 "Personal Studio Rack Series" effects from one AC109. Please note that the AC109 and DC4 combination are rated at 200mA current capacity and exceeding this may damage the effects, or effect their performance.



PSR6 RACK MOUNT

The PSR6 rack mount will let you mount up to 6 Personal Studio Rack series units.

6 SYSTEM HOOKUP

■ FOR GUITAR

GUITAR



COMPRESSOR CP200



DISTORTION MS10 etc.



EQUALIZER
GE215, GE131 OR PQ104



DELAY
DD100 OR DD200



AMP, MIXER
RM80 · RCM804
OR MULTI TRACK RECORDER

■ FOR BASS

BASS



COMPRESSOR CP200



EQUALIZER
GE215, GE131 OR PQ104



DELAY
DD100 OR DD200



AMP, MIXER
RM80 · RCM804
OR MULTI TRACK RECORDER

■ FOR KEYBOARD

KEYBOARD



EQUALIZER
GE215 OR GE131



DELAY
DD100, DD200



REVERB SDR1000



AMP, MIXER
RM80 · RCM804
OR MULTI TRACK RECORDER

■ FOR TWO INSTRUMENTS

GUITAR

KEYBOARD



CH. A

CH. B

COMPRESSOR CP200

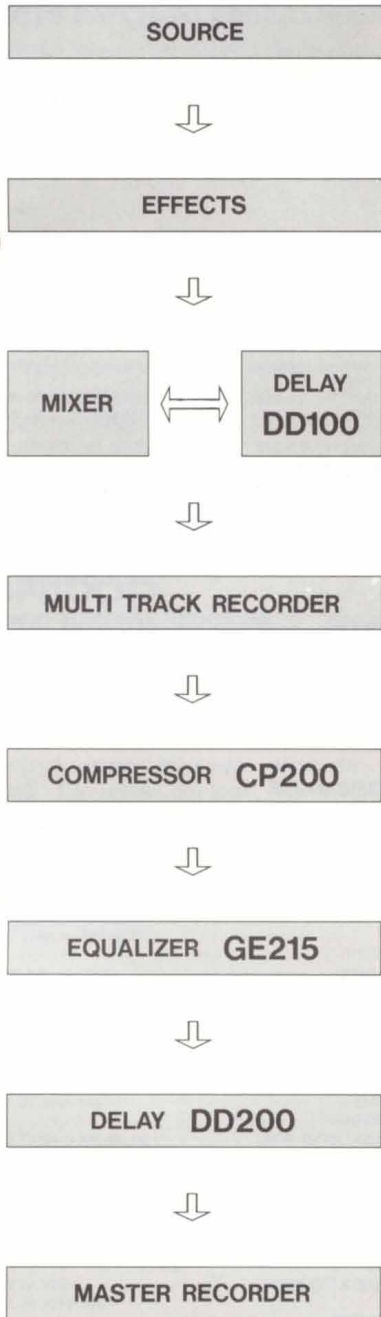


EQUALIZER GE215

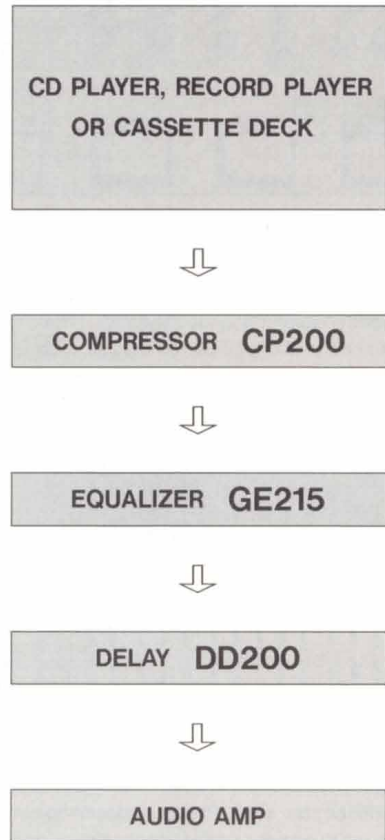


DELAY DD200

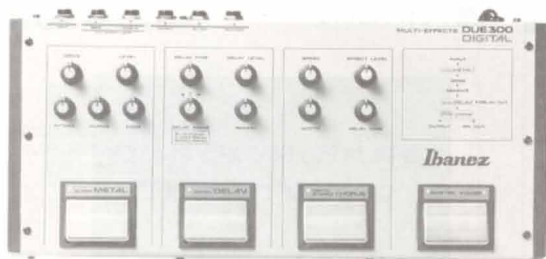
■ FOR PERSONAL RECORDING



■ FOR AUDIO



IBANEZ LINE UP



DUE300 DIGITAL MULTI EFFECTS

Ibanez updates its popular floor system effects format with three of the hottest sounds available. The Digital Delay (with Hold) and Digital Stereo Chorus, both using the exclusive IDPC conversion process, are combined with today's most desired overdrive: the Super Metal Distortion. Together they form the most powerfully versatile trio of effects for guitarists, bassists and keyboard players.

■ SPECIFICATIONS

- SUPER METAL DISTORTION
 - MAXIMUM GAIN 62 dB
- DIGITAL CHORUS
 - DELAY TIME 1 msec TO 8 msec
 - BANDWIDTH 12 kHz
 - SWEEP SPEED 0.6 Hz TO 6 Hz
 - EQUIVALENT INPUT NOISE -97 dBV(IHF-A INPUT SHORTED)
- DIGITAL DELAY
 - DELAY TIME 28 msec TO 224 msec, 224 msec TO 1800 msec
 - HOLD TIME 224 msec TO 1800 msec
 - BANDWIDTH 7 kHz
 - EQUIVALENT INPUT NOISE -95 dBV(IHF-A INPUT SHORTED)
- OVERALL
 - INPUT IMPEDANCE 500 kohms(INPUT/EFFECTS RECEIVE)
 - OUTPUT IMPEDANCE LESS THAN 1 kohms(OUTPUT/INV. OUTPUT/DELAY OUTPUT)
 - DIMENSION 185(W) x 70(H) x 390(D)mm
 - WEIGHT 2.8 kg
 - POWER REQUIREMENT 120V AC/60 Hz 6W 220V AC/50 Hz 7W 240V AC/50 Hz 7W



DUE400 DIGITAL MULTI EFFECTS

The DUE400 Digital Multi-Effects System combines four of the most popular professional effects and an external effects loop with an amazingly versatile PROGRAMMABLE patching system that lets you preset and recall 128 separate system patches – instantly! The DUE400 incorporates four effects: Compressor/Limiter, Super Metal Distortion, Digital Chorus/Flanger and Digital Delay. Each effect features the latest advances by Ibanez engineers, and provide professional quality sounds unmatched in the industry today. The external effects loop, master level control and assignable second output give the DUE400 the versatility to fit into (and control) any signal processing system, in live, studio or practice situations.

■ SPECIFICATIONS

- COMPRESSOR
 - COMPRESSION RANGE 45 dBV TO -10 dBV
 - MAXIMUM COMPRESS MAXIMUM COMPRESSION 35 dB
- SUPER METAL DISTORTION
 - MAXIMUM GAIN 62 dB
- DIGITAL CHORUS/FLANGER
 - DELAY TIME CHORUS = 2 msec TO 16 msec
FLANGER = 1 msec TO 8 msec
 - BANDWIDTH 12 kHz
 - SWEEP RATIO CHORUS = 0.6 Hz TO 5 Hz
 - SWEEP SPEED FLANGER = 1.8 Hz TO 11 Hz
 - EQUIVALENT INPUT NOISE -90 dBV(IHF-A INPUT SHORTED)
- DIGITAL DELAY
 - DELAY TIME 28 msec TO 224 msec
224 msec TO 1800 msec
 - HOLD TIME 224 msec TO 1800 msec
 - BANDWIDTH 7 kHz
 - EQUIVALENT INPUT NOISE -90 dBV(IHF-A INPUT SHORTED)
- OVERALL
 - INPUT IMPEDANCE 100 kohms(INPUT/EFFECTS RECEIVE)
 - OUTPUT IMPEDANCE LESS THAN 1 kohms
(OUTPUT/EFFECTS SEND/SUB OUTPUT)
 - MASTER LEVEL CONTROL 0 TO 0 dB
 - DIMENSION 482(W) x 265(D) x 44(H)mm
 - WEIGHT 3.7 kg
 - POWER REQUIREMENT 120V AC 60 Hz 19W 220V/240V AC 50 Hz 21W
- MIDI (REMOTE)
 - PROGRAM NUMBER 0 TO 127
 - MIDI CHANNEL 1 ch TO 16 ch



EPP400 EFFECTS PATCHING PROGRAMMER

The EPP400 Effects Patching Programmer from Ibanez answers the long-standing need of musicians and studios using multiple processor systems: to organize signal processors to be used in any order, and to change the sound instantaneously. This is the most powerful way to use your effects, with total flexibility and control.

The EPP400 lets you arrange up to five separate effects loops in any order you require, turning any of them on or off. Your own personal effects "patches" can be memorized for later recall. The EPP400 will remember up to 128 different patches, each of which may be edited at any time. A second "stereo" output can be assigned to any loop, and a choice of two inputs and two outputs may also be selected for recall.

■ SPECIFICATIONS

- EFFECTS LOOP
 - LOOP 1 SEND/RECEIVE
 - LOOP 2 SEND/RECEIVE
 - LOOP 3 SEND/RECEIVE/SUB RECEIVE
 - LOOP 4 SEND/RECEIVE/SUB RECEIVE
 - LOOP 5 SEND/RECEIVE/SUB RECEIVE
- INPUT IMPEDANCE 100 kohms
 - INPUT 1 & 2
 - RECEIVE (LOOP 1 TO 5)
 - SUB RECEIVE (LOOP 3 TO 5)
- OUTPUT IMPEDANCE LESS THAN 1 kohms
 - OUTPUT 1 & 2
 - SUB OUTPUT
 - SEND (LOOP 1 TO 5)
- MAXIMUM INPUT LEVEL +18 dBV
- EQUIVALENT INPUT NOISE -94 dBV(IHF-A INPUT SHORTED)
- MIDI (REMOTE)
 - PROGRAM NUMBER 0 TO 127
 - MIDI CHANNEL 1 ch TO 16 ch
- DIMENSION 482(W) x 265(D) x 44(H)mm
- WEIGHT 3.4 kg
- POWER REQUIREMENT 120V AC OR 220-240V/50-60 Hz 14W



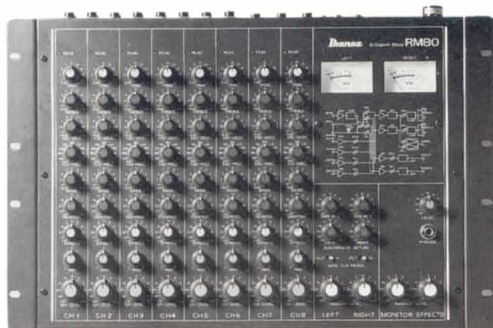
SDR1000 STEREO DIGITAL REVERB

The Ibanez SDR1000 Stereo Digital Reverb redefines digitally-simulated reverberation with true stereo operation, 16-bit digital processing for crystal-clear reverberation, and factory presets programs developed by professional studio engineers, especially for the SDR1000. The results are room, hall and plate simulations of unusual naturalness and clarity. The SDR1000 also provides "gated" and "reverse" reverb effects, as well as dual multi-tap delay (echo) processing.

The SDR1000 is truly a digital processor for everyone.

■ SPECIFICATIONS

- FACTORY PROGRAM (ROM) 30
- USER PROGRAM (RAM) 70
- PARAMETERS REV. T/PRE-DELAYE. REF. T/E. REF. LEV/RT. HIGH/ SIZE/GATE T/F. B. LEV/TAP/DEPTH/SPEED/EFF. LEV EQ LOW/L. MID/H. MID/HIGH MIDI PROG. NO/MEMORY/MIDI CH
- REMOTE CONTROL
 - MEMORY UP PHONE JACK (FS1M)
 - MEMORY DOWN PHONE JACK (FS1M)
 - EFFECT ON/OFF PHONE JACK (FS1M)
 - HOLD ON/OFF PHONE JACK (FS1M)
 - REMOTE DIN 6P (IFC 60)
 - MIDI IN DIN 5P
- MIDI REMOTE CONTROL PROGRAM CHANGE OMNI ON/OFF
- DISPLAYS
 - FL 8 DIGIT, 8 SEGMENT
 - LED INPUT LEVEL INDICATOR/STEREO 7 SEGMENT/CHANNEL LED 5 FUNCTION KEYS, 14 PARAMETER KEYS, AND BYPASS
- LEVEL CONTROL INPUT LEVEL CONTROL/STEREO DRY DIRECT SIGNAL OUTPUT LEVEL CONTROL
- EFFECT PROCESSED SIGNAL OUTPUT LEVEL CONTROL
- BYPASS ON/OFF (POWER OFF: BYPASS ON)
- FREQUENCY RESPONSE
 - DIRECT 20 - 20 kHz
 - EFFECT 20 - 10 kHz
- DYNAMIC RANGE MORE THAN 90 dB
- TOTAL HARMONIC DISTORTION LESS THAN 0.03%
- DIGITAL CODING 16 bit LINEAR PCM
- SAMPLING RATE 26 kHz
- EQUALIZER
 - LOW 100 Hz -12 ~ +12 dB
 - L. MID 400 Hz -12 ~ +12 dB
 - H. MID 1.6 kHz -12 ~ +12 dB
 - HIGH 6.4 kHz -12 ~ +12 dB
- AUDIO INPUT PHONE JACK x 2, RCA PIN JACK x 2
- INPUT LEVEL 20, +4 dBV
- INPUT IMPEDANCE 47 kohms
- AUDIO OUTPUT PHONE JACK x 2, RCA PIN JACK x 2
- OUTPUT LEVEL 20, +4 dBV
- OUTPUT IMPEDANCE 600 ohms
- POWER 80 Hz 120V AC 50, 60 Hz 220V AC 30 W
- DIMENSION 482(W) x 44(H) x 320(D)mm
- WEIGHT 5.5 kg



RM80 8CH STEREO MIXER

Ibanez brings you the latest in mixing board technology. Never before have so many features been put into a rack mount mixer. Features like 8 inputs, stereo outputs, individual monitor and effects sends, parametric EQ, low cut filters, VU metering, RCA tape inputs and outputs. Headphone monitoring and more! Size, features and price combine to create the perfect mix.

■ SPECIFICATIONS

- INPUT CHARACTERISTICSMIC INPUT 47 kohms, -50 dBv
CH. RETURN 47 kohms, -20 dBv
EFFECTS RETURN 47 kohms, -20 dBv
- OUTPUT CHARACTERISTICSCH. SEND 10 kohms, -20 dBv
MASTER OUT 10 kohms, +4 dBv
MONITOR OUT 10 kohms, +4 dBv
EFFECTS OUT 10 kohms, -20 dBv
LINE OUT 10 kohms, -20 dBv
PHONE 8 ohms, 0.25W x 2
- FREQUENCY RESPONSE30 Hz - 20 kHz (±2 dB)
- T.H.D. (1 kHz)0.03%
- HUM AND NOISEEQUIVALENT INPUT NOISE -121 dB (IHF-A)
- MAXIMUM GAINMIC INPUT/MASTER OUT 60 dB
MIC INPUT/MONITOR SEND 60 dB
MIC INPUT/EFFECTS SEND 42 dB
MIC INPUT/CH. SEND 33 dB
MIC INPUT/LINE OUT 40 dB
CH. RETURN/MASTER OUT 27 dB
EFFECT RETURN/MASTER OUT 30 dB
TAPE IN/MASTER OUT 30 dB
- MAXIMUM INPUT LEVELTrim 0 dB +18 dBv
Trim 30 dB -12 dBv
- MAXIMUM OUTPUT LEVEL1 kHz, 600 ohms LOAD +18 dBv
- EQUALIZATIONBASS ±15 dB AT 100 Hz
TREBLE ±15 dB AT 10 kHz
- PARAMETRIC EQMIDDLE ±15 dB
MID. FREQUENCY 200 Hz TO 5.6 kHz
- VU MASTER0 VU=4 dBv
- LED INDICATOR (CH. SEND LEVEL)RED +15 dBv
- POWER REQUIREMENT120V AC 60 Hz 19W
220V -240V AC 50 Hz 19W
- DIMENSION482(W) x 95(H) x 320(D)mm
- WEIGHT5.7 kg

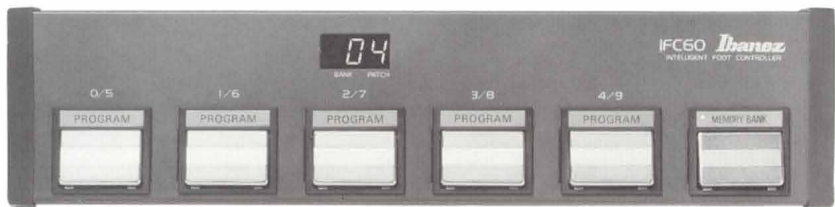


RCM804 8CH 4 OUT MIXER

The Ibanez RCM804 is a versatile 8-in by 4-out recording mixer that is capable of performing all of your recording needs for years to come. It features 8 inputs with 3 band equalization, 1/4" and RCA type inputs, and submixing capabilities for either 2 or 4 channel outputs. The RCM804 features quality found in mixers costing much more.

■ SPECIFICATIONS

- FREQUENCY RESPONSE20 Hz - 20 kHz (-3, +0.5 dB/+4 dBv)
- EQUIVALENT INPUT NOISE124 dB (IHF-A)
- T.H.D.LESS THAN 0.3% (20 Hz - 20 kHz +10 dBv GAIN MAX)
- MAXIMUM INPUT LEVEL+18 dBv (TRIM 0 dB)
-22 dBv (TRIM 40 dB)
- MAXIMUM OUTPUT LEVEL+17 dBv (600 ohms LOAD,
LESS THAN 1% T.H.D.)
- MAXIMUM GAINMIC INPUT/MASTER OUT 70 dB
MIC INPUT/MONITOR OUT 70 dB
MIC INPUT/EFFECT OUT 70 dB
MIC INPUT/CUE OUT 72 dB (MONO SELECT)
MIC INPUT/SEND 43 dB
SEND/MASTER OUT 29 dB (MONO SELECT)
AUX INPUT/MASTER OUT 30 dB (MONO SELECT)
- EQUALIZER CHARACTERISTICSBASS ±15 dB (100 Hz SHELVING)
MID ±15 dB (2.5 kHz PEAKING)
TREBLE ±15 dB (10 kHz SHELVING)
- CROSSTALK (1 kHz MAX. GAIN)INPUT CH TO INP
LESS THAN -
- INPUT CONTROLTRIM 40 dB VALIABLE
TREBLE 10 kHz ±15 dB
MID 2.5 kHz ±15 dB
BASS 100 Hz ±15 dB
MONITOR VR
EFFECT VR
CUE
OUTPUT SELECT (1-4)
PAN
CH FADER
- AUX IN CONTROLAUX IN VR (1-4)
OUTPUT SELECT (1-4)
- OUTPUT CONTROLMASTER FADER (1-4)
MONITOR FADER
EFFECTS FADER
CUE
- PEAK INDICATORLIGHTS +15 dB (6 dB BELOW CLIPPING)
- METERFIP BAR GRAPH METER (MASTER 1-4)
OVU= +4 dBv
- DIMENSION480(W) x 110(H) x 310(D)mm
- WEIGHT5.7 kg



**IFC60
INTELLIGENT
FOOT CONTROLLER**

The IFC60 Intelligent Foot Control System provides complete footswitch control of the DUE400 and EPP400. Select any of the 128 programs from a remote location, without interrupting your playing.



**MFC60
MIDI
FOOT CONTROLLER**

The Ibanez MFC60 MIDI FOOT CONTROLLER allows footswitch selection of the 128 MIDI patch programs. The MFC60 features 6 footswitches, a 3-digit LED readout, and may transmit on any MIDI channel. The MFC60 adds a new dimension of dynamic control for MIDI musicians.



**MIU8
MIDI INTERFACE UNIT**

The MIU8 can be connected to up to 8 different MIDI equipped devices and control all of them from the main MIDI controller. The MIU8 can also be connected to the Ibanez IFC60 Foot Controller to change the MIDI program number of up to 8 MIDI equipped devices from a convenient remote location.



**FS1M
1CH FOOT
SWITCH
(momentary
type)**



**FS1L
1CH FOOT
SWITCH
(alternately
type)**

