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.
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**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 at 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 changes 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 has occurred. 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 invermix 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, chorusing, 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. Patterting (in series) one channel into the other for 2 effects simultaneously.
3. Pattern (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).

**USE**

### DD100

1. **SUBLTRE 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.

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.

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.

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.

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.

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.

   ![Stereo Echo Diagram]

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, creating the perception of a random pattern of echos.

   ![Multi-Tap Delay Diagram]

3. **CHORUS & DELAY**

In example 3, Channels A and B operate very differently. A doing chorus while B does delay. When one instrument is used and footswitches are hooked up to both units, either effect is available and both can be combined along with the dry sound, for a total of 4 "presets" easily footswitched live. In the second patching example the two effects can be used simultaneously on different instruments.

   ![Chorus & Delay Diagram]

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.

   ![Stereo Ping-Pong Diagram]
### DELAY TIME CHART

#### DD100

<table>
<thead>
<tr>
<th>NOTE</th>
<th>TEMPO</th>
<th>DELAY TIME</th>
<th>DELAY TIME</th>
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<tr>
<td></td>
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<td>214ms</td>
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#### DD200 CH. A

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<td>187ms</td>
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### SPECIFICATIONS

#### DD200

<table>
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<th>DD100</th>
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<tr>
<td><strong>DELAY TIME</strong></td>
<td>1: 10 ms = 64 ms</td>
</tr>
<tr>
<td>2: 20 ms = 256 ms</td>
<td></td>
</tr>
<tr>
<td>3: 30 ms = 1024 ms</td>
<td></td>
</tr>
<tr>
<td>4: 40 ms = 4096 ms</td>
<td></td>
</tr>
<tr>
<td><strong>MODULATION SWEEP SPEED</strong></td>
<td>0.00 Hz - 9 Hz</td>
</tr>
<tr>
<td><strong>MODULATION SWEEP RATIO</strong></td>
<td>0.06 Hz - 13 Hz</td>
</tr>
<tr>
<td><strong>FREQUENCY RESPONSE</strong></td>
<td>20 Hz - 12 kHz (±3 dB)</td>
</tr>
<tr>
<td><strong>TOTAL HARMONIC DISTORTION</strong></td>
<td>0.01%</td>
</tr>
<tr>
<td><strong>TOTAL INPUT NOISE</strong></td>
<td>100 dB (ref: A INPUT SHORTED)</td>
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<tr>
<td><strong>INPUT IMPEDANCE</strong></td>
<td>1 kohm ± 10 dB (±10 dBv)</td>
</tr>
<tr>
<td><strong>OUTPUT IMPEDANCE</strong></td>
<td>1 kohm ± 10 dB (±10 dBv)</td>
</tr>
<tr>
<td><strong>CURRENT CONSUMPTION</strong></td>
<td>20 mA (max)</td>
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<td><strong>DIMENSIONS</strong></td>
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<td><strong>WEIGHT</strong></td>
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#### DD200

<table>
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<td><strong>DELAY TIME</strong></td>
</tr>
<tr>
<td>2: 20 ms = 16 ms</td>
</tr>
<tr>
<td>3: 30 ms = 64 ms</td>
</tr>
<tr>
<td>4: 40 ms = 256 ms</td>
</tr>
<tr>
<td><strong>MODULATION SWEEP SPEED</strong></td>
</tr>
<tr>
<td><strong>MODULATION SWEEP RATIO</strong></td>
</tr>
<tr>
<td><strong>FREQUENCY RESPONSE</strong></td>
</tr>
<tr>
<td><strong>TOTAL HARMONIC DISTORTION</strong></td>
</tr>
<tr>
<td><strong>TOTAL INPUT NOISE</strong></td>
</tr>
<tr>
<td><strong>INPUT IMPEDANCE</strong></td>
</tr>
<tr>
<td><strong>OUTPUT IMPEDANCE</strong></td>
</tr>
<tr>
<td><strong>CURRENT CONSUMPTION</strong></td>
</tr>
<tr>
<td><strong>DIMENSIONS</strong></td>
</tr>
<tr>
<td><strong>WEIGHT</strong></td>
</tr>
</tbody>
</table>
FEATURES AND PRINCIPLES

The GE131 and GE215 equalizers allow the user to see a "graphic" representation of the frequency response curve which can be imposed 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 3/1 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 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. A line and a zero position are provided for reference to the flat or zero voltage 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 A.P. system or microphone on a monitor talkback 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 subjective equalization at the problem frequency is necessary. In audio monitoring applications, equalization can help attain a flat playback system when the speaker or room is reproducing or reflecting an even response throughout the audio spectrum.

USE

GE215

1. STEREO SOUND SHAPING

In example 1, the GE215 is used to generally shape the tone of a stereo sound system. Gradual reduction of low frequencies helped to "clean up" a muddy sounding room. The drastic reduction at 250 Hz was necessary because of 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 10K.

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 around 300 - 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 was cut off at 18KHz. Overall level was increased to counteract all the subtractive eq.

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 (± 1dB)
ALL EQUALIZATION CONTROLS AT FLAT
EQUIVALENT INPUT NOISE: -105 dbm
20 dBm (0.01% of full range)
MAXIMUM INPUT LEVEL: 210 dbm
INPUT IMPEDANCE: 1 Mohm, 25 pF
EQUALIZATION CONTROL LIMITS:
CENTER FREQUENCY 1 kHz (MAX)
25, 40, 80, 160, 320, 640, 1280, 2560 kHz
RANGE OF BOOST - 12 dB
PEAK INDICATOR:
TURN ON WHEN THE OUTPUT LEVEL REACHES +4 dB
CURRENT CONSUMPTION: 70mA (+12V 3V)
DIMENSION: 482(W) x 132(D) x 44(H)mm
WEIGHT: 1.4kg

GE215

FREQUENCY RESPONSE: 20 Hz - 20 kHz (± 1dB)
8 - 30 dB (ALL EQUALIZATION CONTROLS AT FLAT)
EQUIVALENT INPUT NOISE: -105 dbm
20 dBm (0.01% of full range)
MAXIMUM INPUT LEVEL: 150 dbm
INPUT IMPEDANCE: 1 Mohm, 25 pF
EQUALIZATION CONTROL LIMITS:
CENTER FREQUENCY 1 kHz (MAX)
25, 40, 80, 160, 320, 640, 1280, 2560 kHz
RANGE OF BOOST - 12 dB
PEAK INDICATOR:
TURN ON WHEN THE OUTPUT LEVEL REACHES +4 dB
CURRENT CONSUMPTION: 70mA (+12V 3V)
DIMENSION: 482(W) x 132(D) x 44(H)mm
WEIGHT: 1.4kg
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 an expensive compressor-limiter, 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 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 60dB change at the input would be only 30dB different at the output. Hard limiting or trim transients is achieved at the highest setting. The output control is used to maintain unity gain 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 gate opening 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 the various and the compressor can be variable, and are automatically adjusted following the input level.

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 bypassed, 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/processor 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 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 subjected to compression (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.

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 a setting on the gate ratio and threshold will result in an unnatural release of the note.

Gating alone as 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 remove a non-linear decay to a heavily reverberated 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 noise before and after the desired sound won't be sampled along with it. This makes for a cleaner samples that is easier to work with. Some compression or gating can also help get the best sample. Loss of dynamic character can be re-synthesized on most samplers via the envelope generators.


**FEATURES AND PRINCIPLES**

The PQ104 equalizer gives the user 9 bands of control in four primary areas: gain, low, mid, and high. Each section is adjustable in 2 octaves (2 octaves on each side of the center frequency). The bandwidth shapes the affected area. A center frequency is set by the 3rd parameter, which is the gain control. There is a 1500Hz of boost or cut available. Four bands of parametric EQ give plenty of control over most audio signals but just in case you need them, two extra bands are provided. These controls have preset frequency bands of 100Hz and 10kHz called bass and treble. Unlike the parametric, the bands are preset to gradually increase in amplitude as the center frequency is increased, then reduced and continue the effect throughout the majority 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 because of the sensitivity of the system to noise, the high frequencies of the equalizer is provided. An input level control with LED indicators allows setting of the incoming signal's level. The level that is optimum for the equalizer can be set. Once adjustments to the equalizer are made, the levels may need to be re-adjusted 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 overpower any changes in volume caused by extreme equalization. Comparison of the original signal is quickly obtained by using the input/output switch.

Another circuit that behaves like a HPF (high pass filter) provides more equalization by passing all frequencies above 40kHz, where many sub-harmonic noises reside that are harmful to speakers and unnecessary to musical signals. When using the PQ104 for tame shaping, a wider bandwidth setting provides a smoother sound, 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 whole range 2 of the five equalizer sections when extreme feedback problems occur at one frequency. When shaping the sound 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.

**SPECIFICATIONS**

- **INPUT IMPEDANCE**: 47 kOhms
- **OUTPUT IMPEDANCE**: LESS THAN 10 kOhms
- **FREQUENCY RESPONSE**: 20 Hz - 20 kHz (Flat)
- **EQUIVALENT INPUT NOISE**: -55 dB (A-Weighted)
- **CENTER FREQUENCY**: LOW: 20 Hz - 2 kHz
- **HIGH-MID**: 20 kHz - 5 kHz
- **HIGH**: 100 kHz - 10 MHz
- **BANDWIDTH**: LOW: 20 Hz - 2 kHz
- **HIGH-MID**: 10 kHz - 15 kHz
- **HIGH**: 15 kHz - 20 kHz
- **GAIN LIMIT**: 10 dB
- **INPUT LEVEL**: 1000 V
- **FOOTSWITCH**: 2000 V
- **POWER**: 100 W
- **INDICATOR**: 100 W
- **WEIGHT**: 0.5 kg

**4 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 mixed 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. Preemphasis of highs when recording on a tape recorder or sampler known to roll off high end.
6. Filtering of frequencies that cause aliasing noise in samplers.
7. Additive and subtractive assistance for creating sounds on synthesizers.

**FEEDBACK CONTROL**

Feedback control, as in the extreme case of example 3. is one of the most useful applications of the PQ104. A very prominent roar was so loud in this sound system that, even though the frequency was found by cutting the low-end, it took both the low and 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 subtration of mids solved this problem. A squeezing high frequency was subtrated and boosted added on the treble tone control to counterbalance.

**VOCAL FOR RECORDING**

Example 1 shows the setting after a particular vocalist's microphone was set for recording purposes. The low band added warmth, while the low-mid bands 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.
**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.

---

**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
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 powfully versatile trio of effects for guitarists, bassists and keyboard players.

**SPECIFICATIONS**
- **Input Impedance:** 500 kohm
- **Output Impedance:** Less than 1 ohm
- **Dimension:** 220 x 165 x 79 mm
- **Weight:** 2.5 kg
- **Power Requirement:** 30 W
- **MIDI Number:** 107
- **MIDI Channel:** 1 to 16

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**
- **Input Impedance:** 500 kohm
- **Output Impedance:** Less than 1 ohm
- **Dimension:** 220 x 165 x 79 mm
- **Weight:** 2.5 kg
- **Power Requirement:** 30 W
- **MIDI Number:** 107
- **MIDI Channel:** 1 to 16

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 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 "shared" output can be assigned to any loop, and a choice of two inputs and two outputs may also be selected for recall.

**SPECIFICATIONS**
- **Input Impedance:** 500 kohm
- **Output Impedance:** Less than 1 ohm
- **Dimension:** 12 x 15 x 14 cm
- **Weight:** 3.5 kg
- **Power Requirement:** 250 VAC or 220-240VAC @ 60 Hz

SDR1000
Stereo Digital Reverb

The Ibanez SDR1000 Stereo Digital Reverb redifines digitally-simulated reverb and stereo operation, 16-bit digital processing for crystal-clear reverb, and factory presets programs developed by professional audio 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**
- **Input Impedance:** 500 kohm
- **Output Impedance:** Less than 1 ohm
- **Dimension:** 12 x 15 x 14 cm
- **Weight:** 3.5 kg
- **Power Requirement:** 250 VAC or 220-240VAC @ 60 Hz
**RMBO 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**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Specification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mic input</td>
<td>47 kohms, -50 dB</td>
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<tr>
<td>Ch. return</td>
<td>147 kohms, -20 dB</td>
</tr>
<tr>
<td>Output characteristics</td>
<td>-CH. SEND</td>
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<tr>
<td>Output level</td>
<td>10 kohms, +4 dB</td>
</tr>
<tr>
<td>Effects return</td>
<td>-CH. SEND</td>
</tr>
<tr>
<td>Line out</td>
<td>10 kohms, +4 dB</td>
</tr>
<tr>
<td>Phone</td>
<td>50 ohms, 0.25V x 2</td>
</tr>
<tr>
<td>Frequency response</td>
<td>0-10000 Hz, 0.1 dB</td>
</tr>
<tr>
<td>THD (1 kHz)</td>
<td>1%</td>
</tr>
<tr>
<td>Hum and noise</td>
<td>-120 dB (F60-Hz)</td>
</tr>
<tr>
<td>Maximum gain</td>
<td>16 channel</td>
</tr>
<tr>
<td>Mic input/master output</td>
<td>60 dB</td>
</tr>
<tr>
<td>Mic input/mic level output</td>
<td>60 dB</td>
</tr>
<tr>
<td>Mic input/effects send</td>
<td>42 dB</td>
</tr>
<tr>
<td>Mic input/line output</td>
<td>42 dB</td>
</tr>
<tr>
<td>Effect return/master output</td>
<td>30 dB</td>
</tr>
<tr>
<td>Tape input/master output</td>
<td>30 dB</td>
</tr>
<tr>
<td>Maximum input level</td>
<td>-10 dB</td>
</tr>
<tr>
<td>Input level</td>
<td>1000 mV rms, 30 dB</td>
</tr>
<tr>
<td>Equalizer characters</td>
<td>-10 dB</td>
</tr>
<tr>
<td>Parametric EQ</td>
<td>MD</td>
</tr>
<tr>
<td>Mic frequency 200 Hz to 20 kHz</td>
<td>HMD</td>
</tr>
<tr>
<td>LED indicator (CH. SEND level)</td>
<td>RED</td>
</tr>
<tr>
<td>Power requirement</td>
<td>120W AC 60 Hz, 150V</td>
</tr>
<tr>
<td>Dimensions</td>
<td>430x320x220 mm</td>
</tr>
<tr>
<td>Weight</td>
<td>5.7 kg</td>
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</tbody>
</table>

**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**

<table>
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<tr>
<th>Feature</th>
<th>Specification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Frequency response</td>
<td>20 Hz - 20 kHz, -3 dB, 0.5 dB, -4 dB</td>
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<tr>
<td>Equivalent input noise</td>
<td>124 dB, 0.5 dB, 4 dB</td>
</tr>
<tr>
<td>THD (1 kHz)</td>
<td>1%</td>
</tr>
<tr>
<td>Hum and noise</td>
<td>-120 dB (F60-Hz)</td>
</tr>
<tr>
<td>Maximum gain</td>
<td>16 channel</td>
</tr>
<tr>
<td>Mic input/master output</td>
<td>60 dB</td>
</tr>
<tr>
<td>Mic input/mic level output</td>
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</table>

**IFC60 INTELLIGENT FOOT CONTROLLER**

The IFC60 Intelligent Foot Control System provides complete footswitch control of the DUE400 and EP400. 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.

**MIUB MIDI INTERFACE UNIT**

The MIUB can be connected to up to 8 different MIDI equipped devices and control all of them from the main MIDI controller. The MIUB 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.

All Specification subject to change without notice or obligation.