Because the spring delay lines are located in the same chassis as the electronics it may be inconvenient to install the reverb away from hum fields. Therefore, considerable attention has been given to hum-shielding the 111B. The spring pickup coils are protected with added mu-metal shields and the steel case of the 111B provides increased protection.

If the Model 111B is used with a recording studio-type mixing console, it is connected to the echo send and echo return busses in the customary manner. For users who wish to use the 111B without such a console, an auxiliary output containing a mixture of direct sound and reverberated sound is available. The amount of reverberated sound is adjustable with the front panel Output Atten control.

Warranty and Service

Orban's use of top quality parts, industrial-quality construction, and special test and burn-in procedures makes it highly unlikely that a user will ever experience any trouble with a 111B reverb. However, it's nice to know that the 111B is protected by a one-year parts and labor warranty, and that Orban is well-known for its fast, reasonable-cost service. Installation and "in-house" troubleshooting are made easy by an outstanding instruction manual which includes detailed installation instructions, performance verification tests, circuit description, troubleshooting hints, alignment instructions, and schematic diagram, as well as the user operator's instructions.

When shopping for a reverb, alway consider parts quality, workmanship, reliability, warranty, service, and manual in addition to the obvious features. While the Orban 111B's features are outstanding, its true value lies in assuring **all** aspects of owner satisfaction. Compare before you buy.

It's a proven performer with the right sound at the right price.

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San Francisco, CA 94107
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Telex: 17-1480

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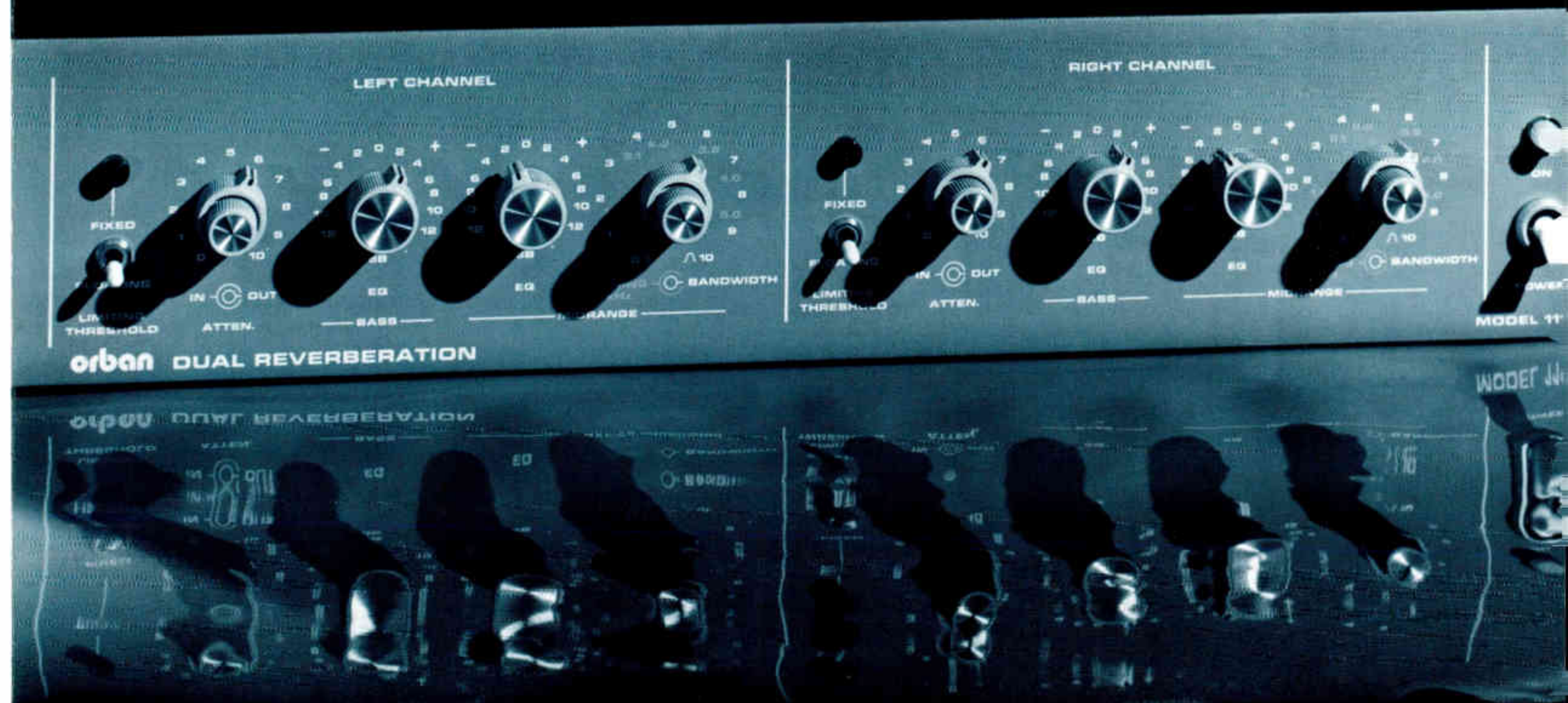
SPECIFICATIONS

Number of Channels: two, entirely independent except for power supply.
Reverberation Element: six-spring array (per channel).
Frequency Response: See Fig. 1.
Decay Time: See Fig. 1.
Delay Time: Approximately 30 milliseconds between direct sound and first reflection.
Input Level: will accept input levels between −30 and +4 dBm. Audio-taper **Input level** attenuator available on the front panel. Limiter will control overloads up to 25 dB above limiting threshold before clipping and distortion occur.
Input Impedance: 10,000 ohms, unbalanced. Source impedance non-critical.
Output Level: nominally 0 dBm, adjustable by front panel control, ±20 dBm clipping level allows adequate headroom for equalization and spring resonances.
Output Impedance: 600 ohms; transformer-coupled; balanced and floating.
Limiter Attack Time: less than 100 micro-seconds.
Limiter Release Time: Dual time-constant circuit adjusts release time as a function of the program.
Compression Ratio (FIXED Mode): greater than 10:1.
Limiter-Induced Harmonic Distortion (@5 kHz): less than 0.2%.
Limiter Element: Junction Field-Effect Transistor.
Bass Equalizer:
Type: Shelving
Turnover Frequency: 500 Hz.
Equalization Range: ±12 dB, reciprocal.
Midrange Equalizer:
Type: quasi-parametric peaking.
Peaking Frequency: continuously variable, 1.5 to 5.5 kHz.
Equalization Range: continuously variable ±12 dB, reciprocal.
Bandwidth Range: can adjust "Q" from 0.5 to 5.0 with any setting of TUNING control.
Control Interaction: TUNING and EQUALIZATION controls also vary "Q." Otherwise, all controls are independent and non-interacting.
Weighted System Signal/Noise Ratio: better than 76 dB.
Indicators:
POWER ON pilot lamp.
LED automatically lights whenever limiter is in FIXED mode (one per channel).
Audio Connector: Jones 140-Y barrier strip (#5 screw).
Power Connector: "U-Ground" power cord to United States standards.
Power Requirements: 115/230 volt AC ±10%. 50-60 Hz, approximately 10 watts.
Dimensions: 19" (48.3 cm) wide × 3½" (8.9 cm) high × 12" (30.5 cm) deep.
Shipping Weight: 10 pounds (4.54 kg).

The Orban 111B Dual Spring Reverb

A Proven Performer
with the Right Sound at the Right Price.



Performance Highlights

- ☐ Two independent channels with six springs per channel
- ☐ Floating threshold peak limiter protects against "twang" and "boing" noises
- ☐ Bass and quasi-parametric midrange EQ allows coloring of echo return
- ☐ Front panel mixed output control
- ☐ Accepts input levels from – 30 to + 4 dBm
- ☐ Industrial-quality construction and rugged package
- ☐ Extremely low signal-to-noise ratio and distortion

The Reverb with a Track Record

The Orban 111B Dual Spring Reverb represents the refinement of over 12 years of experience with spring reverbs. In that time, thousands of 111B's (and its predecessors) have found their way into recording studios, broadcast facilities, schools and colleges, and sound reinforcement systems. The reason is simple: Orban continues to offer the best price/performance value in spring reverbs, resulting in units with excellent sound quality, easy installation, and an outstanding reliability record.

Features

• LOW FLUTTER is assured by the use of six springs per channel. Compared to the low-priced, consumer-grade competition, the sound is much smoother and better integrated.

• SIGNAL-TO-NOISE ratio is optimized by the use of a special low-noise IC preamp, by added mu-metal hum shielding around the spring pickup coils, and by a specially designed limiter circuit which allows the user to utilize the full headroom available in the system without concern for potential overload and distortion. The result: the effective signal-to-noise ratio of the 111B can be more than 6dB better than some of the high-priced competition.

• "TWANG" AND "BOING" NOISES are greatly reduced by this exclusive limiter when operated in the "floating threshold" mode. This circuit serves to eliminate sudden, sharp changes in level regardless of average level. By exploiting the "masking effect" (which lets the direct sound hide the residual "twangs"), even percussion and guitar can be reverberated without unnatural effects.

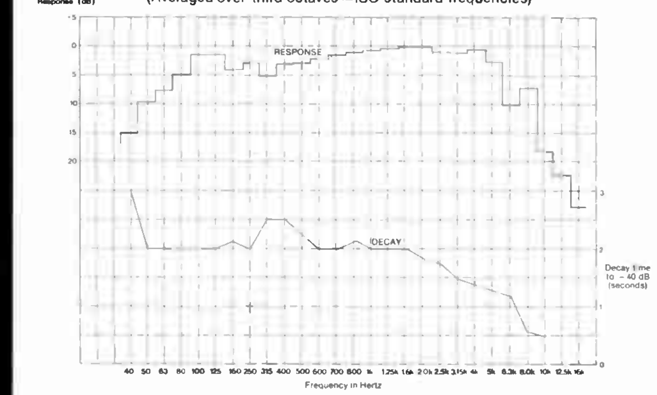
Evaluating a reverb by listening to the echo return alone is essentially meaningless—because reverb is almost never used in this mode. In fact, it is **essential** to evaluate the reverb in a real-world situation (with direct sound mixed in) to perceive the subtle psychoacoustical interaction between the direct sound and the reverb. You will find that the reverberation generator that sounds best when listened to alone may give a totally different impression when direct sound is mixed in.

The 111B was designed with the psychoacoustical interaction between direct and reverberated sound always in mind. Therefore, DECAY VERSUS FREQUENCY does not drop abruptly at high frequencies, unlike some of the high-priced competition. When others tout their naturalness on percussion, watch out—too often, this is achieved at the expense of excessive high frequency damping which gives the highly-audible reverb decay a dull, bassy sound. Because this characteristic is time-varying, it is **not** correctable with fixed equalization.

On the other hand, the 111B's longer high frequency decay results in the bright sound that most pop music demands. In fact, it's the closest you can get to the high-priced "plate" reverb sound in a low-priced reverb. Compare on vocals...strings... guitar...brass. We think that the cost-effective 111B more than stands up to the higher-priced compact spring reverbs on the market.

• FREQUENCY RESPONSE is optimized by means of elaborate fixed equalization in the reverb circuitry. In addition, a bass control and a quasi-parametric midrange equalizer permit the user to tailor the sound to his exact requirements. The versatile midrange equalizer permits continuously variable adjustment of the **frequency** of maximum equalization (1.5 to 5.5 kHz), the **amount** of equalization (up to ± 12dB), and the **bandwidth** (Q's from 0.5 to 5.0). We call it "quasi-parametric because operating the **tuning** control causes the bandwidth to change (unlike our full parametric equalizer, in which the controls are totally non-interacting).

Fig. 1 Frequency Response and Decay Time (to – 40 dB)
(Averaged over third-octaves—ISO standard frequencies)



Complex frequency response and decay time curves effectively simulate natural reverberation.

Installation and Applications

The versatility of the quasi-parametric midrange equalizer complements the simple, inflexible equalization found on many low-cost mixers, and permits the owners of such systems to get the exact reverb sound they want. In addition, the 111B has very high basic input sensitivity (– 30dBm), and a front-panel input gain control makes it usable with all mixers—even those with unusually low level sends.

This versatility is complemented by a fully-professional 0 dBm balanced, floating output. This arrangement vastly improves immunity to RF interference, and assures easy integration into any system without introducing ground loops and hum.

The Orban 245F Stereo Synthesizer

Convincing pseudo-stereo from mono sources.

Performance Highlights

- ☐Creates a convincing pseudo-stereo effect from any mono source
- ☐Total mono/stereo compatibility for FM broadcast applications
- ☐Patented design offers seductive space and depth enhancement
- ☐Saves tracks in multi-track recording situations
- ☐Allows for stereo cart transfers with no phasing problems
- ☐Simple and easy-to-use

Description

The Orban Model 245F Stereo Synthesizer has been designed to take any mono signal and create lifelike pseudo-stereo. Unlike many other techniques, the patented Orban stereo synthesis technique causes no change in spectral balance, does not blur the transient definition, and adds not the slightest audible noise or distortion to the mono original. The stereo output sums back to the original mono for total mono/stereo compatibility. And the simple controls adjust in seconds to create an optimum stereo effect from any mono original. The major new features added to the 245F are a standard active balanced input and provisions for mounting optional output transformers to provide balanced outputs. Enhanced RFI suppression is provided on both audio and power leads. In addition, front-panel cosmetics have been revised.

How it works

The Orban Stereo Synthesizer creates a stereo effect by dividing the mono source signal into five frequency bands. Three of these bands are placed in one stereo output channel; the remaining two are placed in the other channel. The filters are synthesized so that the sum of the two output channels is identical to the mono input. In addition, the sum of the powers in the left and right output channels is equal to the power in the mono input signal, guaranteeing that the stereo will have the same perceived frequency balance as the mono source.

The bandcenters and bandwidths of the midrange bands are adjustable by means of two **dimension** controls, one controlling lower midrange and the other controlling upper midrange. These controls act like frequency-band panpots, and are used to get good left-right channel balance for a given piece of mono source material. With practice, adjustment takes no more than five or ten seconds for a given mono source.

Also provided is a **separation** control which adjusts the level of the stereo difference signal anywhere from zero to the same level as the sum signal. The control is useful for adjusting the audible separation, and also controls the vertical component on a stereo disc or the subchannel modulation (and therefore the stereo and mono loudness) in FM stereo broadcasting. All controls can be adjusted freely throughout their range without fear of losing stereo/mono compatibility.

Recording Studio Applications

Reissuing old mono material

The most obvious application for the Orban Stereo Synthesizer in the recording studio is the reissuing of old mono masters in pseudo-stereo. Because of mono compatibility, this can be done without offending those purists who have been turned off by some of the more bizarre and tasteless pseudo-stereo efforts of the past.

In cutting discs from mono masters, there is no need to go through an added tape generation—the disc can be cut directly through the Stereo Synthesizer.

Dimensionally spreading single tracks in multi-track mixdowns

No matter how many tracks are available on a multi-track recorder, there never seem to be enough. And the first thing to be sacrificed is usually stereo recording of material like drums, strings and horns. All is not lost—mono tracks can be spread in space in the mixdown through the use of the Stereo Synthesizer. Electric or electronic instruments like synthesizer, guitar and organ can be given a sense of space and depth. And the mono input of an echo chamber or artificial reverb generator can be spread in a lifelike way.

Cable TV and Satellite

The 245F is the ideal solution to providing a full-stereo format at the cable headend. When used in conjunction with FM multiplex systems, the 245F is a cost-effective way of providing FM stereo audio from cable audio. The unit can also be used with satellite systems for a similar purpose.

FM Broadcast Applications

Reducing stereo cart phase cancellation

Ever since the advent of the stereo tape cartridge machine, FM stereo broadcasters have been plagued with mono signal degradation due to phase shifts between the two stereo channels. The Orban Stereo Synthesizer can greatly alleviate this problem.

The phase cancellation problem arises because there are usually several frequencies in the high-frequency audio band where the left and right outputs from the stereo cart machine are 180° (or odd multiples thereof) out of phase. At these frequencies, material having equal level on the left and right channels will totally cancel, and at frequencies close to the 180° frequencies, the mono sum will be greatly attenuated.

Because of its filters, the Stereo Synthesizer places most frequencies on the left and right channels with unequal levels. Therefore, even at frequencies where the cart machine is 180° out of phase, cancellation is greatly reduced and the mono sound is notably improved.

The 245F can either be used at the output of a mono cart machine to create a pseudo-stereo effect, or it can be used when transferring material to a stereo cart to reduce phase cancellation due to cart phase problems. In either case, the result is a convincing pseudo-stereo effect with no mono signal degradation.

Stereoizing old mono material, announce mikes, etc.

The Stereo Synthesizer is an ideal way to create a "total stereo" format for AM or FM stereo that includes old LP's, "golden oldies" 45's, agency spots, PSA's and commercials. This material can be recorded on automation tapes without danger of mono phase cancellation. And DJ announcements, live or recorded, can be processed, eliminating the gross inconvenience of stereo-miking the announcer.

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SPECIFICATIONS

Frequency Response of the Stereo Sum Signal:
± 1 dB (re mono input) 20-20,000Hz

Frequency Response of the Sum of the Right and Left Channel Powers:
± 1 dB (re mono power) 20-20,000 Hz

Total Harmonic Distortion: (+19 dBm, 20-20,000 Hz): Less than 0.1%; 0.02% typ.

Noise: (Unweighted, 20-20,000 Hz): less than −80 dBM; −83 dBm typ.

Available Gain: approximately 9 dB (MONO); 14 dB peak (STEREO)

Input: greater than 100 K ohms, balanced bridging. Absolute overload occurs at +26dBm.

Output: approximately 47 ohms unbalanced. Will drive greater than +19 dBm, 20-20,000 Hz into 500 ohms or higher load impedance. Optional transformer-balanced output available.

Input/Output Connector: Type 140-Y barrier strip (#5 screw)

Power Requirements: 115-230 volt 50-60 Hz AC, ±10%, 2 VA. Supplied with "U-Ground" grounding-type plug to United States standards.

Mounting: requires 1¾" (4.5 cm/1unit) of vertical space in an EIA Standard 19" (48.3 cm) rack.

Shipping Weight: 7 pounds (3.2 kg)

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Specifications

Input

Impedance: greater than 10 k ohms active balanced (interfaces with balanced or unbalanced sources).

Level: – 15dBm produces 10dB gain reduction with ATTACK TIME control centered, INPUT ATTEN control fully CW, and RATIO control at infinity-to-one.

Absolute Overload Level: + 21dBm

Output

Impedance: approximately 100 ohms, electronically balanced and floating (drives balanced or unbalanced loads)

Levels: + 4dBm nominal; absolute peak overload better than + 19dBm

Frequency Response

±0.25dB 20-20,000 Hz below limiting threshold

Compressor/Limiter Characteristics

Attack Time: manually adjustable in range of approximately 500us to 200ms; automatically scaled by program content.

Release Time: adjustable in range of approximately 3dB/sec to 80dB/sec; automatically scaled by program content.

Compression Ratio: adjustable from 2:1 to infinity-to-one at threshold. Lower ratios automatically increase as gain reduction increases.

Range of Gain Reduction: from 15dB to 35dB depending on setting of THRESHOLD control; 25dB with THRESHOLD control at center detent.

Total Harmonic Distortion
(ATTACK and RELEASE TIME controls centered, infinite RATIO, 15dB gain reduction): less than 0.05% @ 1kHz. Typically 0.04% @ 100Hz; 0.025% @ 1kHz; 0.035% @ 10kHz.

SMPTE IM Distortion (controls set as above; 60/7000Hz 4:1; 15dB gain reduction): typically 0.2%.

Tracking of Multiple Channels: ±0.5dB.

System Noise

RMS noise in 20-20kHz bandwidth better than 85dB below output clipping threshold for any degree of gain reduction; 90dB typical.

Crosstalk (414A Only)

Better than – 70dB @ 20kHz. Unmeasureable below 10kHz.

Operating Controls

INPUT ATTENUATOR
OUTPUT ATTENUATOR
THRESHOLD
COMPRESSION RATIO
ATTACK TIME
RELEASE TIME
SYSTEM OPERATE/BYPASS (Hardwire Bypass)
STEREO COUPLING (414A Only)
POWER ON/OFF

Indicators

GAIN REDUCTION METER
GAIN REDUCTION OVERLOAD LAMP
OUTPUT CLIP LAMP

Power Requirement

115/230 VAC ±10%; 50-60Hz. U-ground power cord attached.

Dimensions

412A
19″ (48.3cm) wide × 1.75″ (4.5cm/1 unit) high × 5.3″ (13.5cm) deep

414A
19″ (48.3cm) wide × 3.5″ (8.9cm/2 units) high × 5.3″ (13.5cm) deep

Operating Temperature

0-45° C

Warranty

One year, parts and labor. Subject to limitations set forth in our Standard Warranty Agreement.

SPECIFICATIONS SUBJECT TO CHANGE WITHOUT NOTICE.

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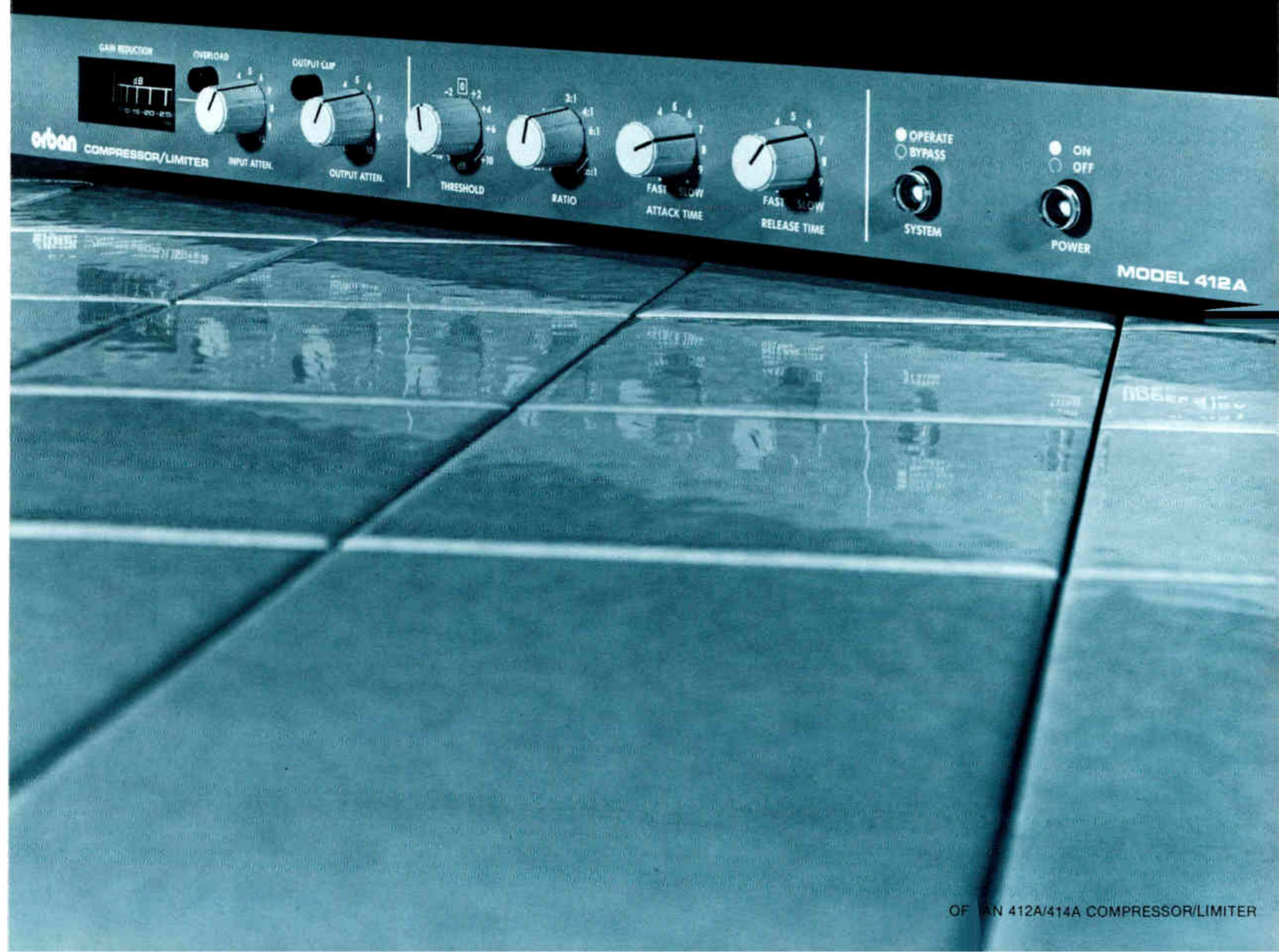
800028-000-01 3/84

The Orban 412A Compressor/Limiter

The Essential AGC: A basic, cost-effective compressor/limiter with remarkably natural sound and extraordinary ease-of-use.



Performance Highlights

☐ Streamlined, straightforward front panel offers the most-demanded user controls, including ATTACK TIME, RELEASE TIME, RATIO, and THRESHOLD. These wide-range controls permit extremely natural sound or special effects.

☐ Exclusive Orban feedback control circuitry (adapted from our popular 424A Gated Compressor/Limiter/De-Esser) achieves remarkably transparent sound.

☐ User controls interact intelligently to simplify and speed setup, and to prevent errors.

☐ Peak limiting and compressor functions are crosscoupled to eliminate potential pumping and modulation effects.

☐ THRESHOLD control with 20dB range allows user to determine the level at which gain reduction first occurs, without changing below-threshold gain. Ideal for sound reinforcement applications.

☐ Proprietary circuitry achieves optimum headroom and signal-to-noise regardless of THRESHOLD control setting.

☐ Front-panel OUTPUT ATTENUATOR control with OUTPUT CLIP LED to indicate line amplifier clipping.

☐ Illuminated, true peak-reading GAIN REDUCTION meter is more accurate and readable than LED displays.

☐ GAIN REDUCTION OVERLOAD lamp warns of control circuit overload due to a demand for G/R which exceeds the range of the VCA.

☐ Hard-wired system bypass switch for fail-safe protection.

☐ Side-chain externally accessible for special effects such as frequency-selective limiting.

☐ Proprietary Class-A Orban VCA features very low distortion and noise.

☐ Stereo 414A has STEREO COUPLING switch to permit either stereo or dual-mono operation; an unlimited number of units can be wire-coupled to track ±0.5dB.

☐ All-metal chassis with RFI suppression on input, output, and AC leads.

Back to Basics

The 412A is Orban's entry into the general-purpose compressor/limiter sweepstakes—it's designed to make **you,** the audio professional, the winner! Based on circuitry from the extremely popular Orban 424A (Gated Compressor/Limiter/De-Esser: the "Studio Optimod"), the 412A offers the controls and features most demanded by audio professionals. This is a no-frills unit with all the essentials—plus no-compromise sound, performance, quality and reliability at a **very** attractive price.

Familiar ATTACK TIME, RELEASE TIME, RATIO and OUTPUT ATTENUATOR controls make operation quick and intuitive. **Both** THRESHOLD and INPUT ATTENUATOR controls are available, so you can adjust the amount of G/R exactly according to the needs of your application—the 412A can keep above-threshold output level **or** below-threshold gain constant, depending on which control you adjust. And Orban's proprietary low-distortion Class-A VCA lets you adjust the THRESHOLD control without compromising headroom or signal-to-noise ratio: They stay optimized over the control's 20dB range.

Attack and release times are program-controlled: the ATTACK TIME and RELEASE TIME controls simply scale the complex time constants faster or slower. With these controls set beyond about "12:00", the 412A produces very natural, open sound with considerable short-term dynamic range—even when the unit is adjusted for high compression ratios. As time constants are sped up, the result is increased loudness and density with a minimum of unnatural compression artifacts. So, depending on the settings of these controls, the 412A can serve as a peak limiter, a pure, gentle compressor, or a combination compressor/limiter.

The unit is utterly simple to operate, and can be used either as a "hands-off" device or as a powerful creative tool. For either use we've built-in an "automatic transmission": The threshold of limiting interacts with both the ATTACK TIME and RATIO controls to keep the peak output approximately constant regardless of control settings. (Of course, it **is** affected by the THRESHOLD and OUTPUT ATTENUATOR controls, as expected.) That way, many annoying corrective readjustments of other controls become unnecessary, making your work easier and more efficient.

This feature provides a bonus: Unlike many other units, it is impossible to severely clip the 412A's VCA unknowingly. Other potential overload conditions are indicated by two LED's: one to indicate program amplifier clipping and the other to indicate overload of the control circuitry. Between the program-controlled time constants, automatic threshold adjustments, and overload monitoring, the 412A is close to foolproof—it facilitates good, fast results even from inexperienced or overburdened operators.

Ultimately, the proof is in the **listening.** We believe that our resolutely un-trendy feedback control circuitry provides a natural sound unmatched at the 412A's modest price—and matched only by our own more sophisticated 424A. Finally, it is possible for smaller studios and production rooms, fixed installations (like churches and theaters), and traveling reinforcement systems, to get top-quality level control at an affordable price. And, because the 412A doesn't compromise basic audio quality, it is fully suited for the most demanding applications requiring its particular assortment of features.

If even more flexibility is needed, consider the 422A (single-channel) and 424A (dual-channel/stereo) Gated Compressor/Limiter/De-Essers, which combine the superb audio quality of the 412A with sophisticated gating circuitry (to prevent compression-induced noise breathing) and an effective de-esser.

Between the 412A and the 422A, Orban offers a compressor/limiter to satisfy virtually any requirement in professional audio—with unmatched quality and singular standards of documentation and customer support.

Order Guide

Model 412A Single Channel Compressor/Limiter

RET-28A XLR-type Connector Field Retrofit Kit

RET-29A TRS Phone Jack Field Retrofit Kit

Model 414A Dual Channel/Stereo Compressor/Limiter

RET-28B XLR-type Connector Field Retrofit Kit

RET-29B TRS Phone Jack Field Retrofit Kit

(Barrier strip connections standard, #5 screw)

Remote Control of the 275A

Remote Control Panel: An optional 19" rack-mount remote control panel provides duplication of all front-panel indicators and functions except for the SEPARATION control.

Control By Automation: The 275A has a rear-panel connector which provides optically-isolated logic inputs for automated control of noise reduction, automatic polarity correction, and synthesis functions. These inputs can accept pulsed (latching) or continuous control signals.

Pulsed signals duplicate the functions of the front-panel buttons. For example, an automated switching system could trigger the desired synthesis mode for each event, setting the 275A's mode with a short contact closure or logic pulse. The 275A would remain in that mode until given another command from the automation or front-panel controls.

Continuous control signals would be used where the desired synthesizer mode is encoded continuously, as in the vertical interval. Continuous control signals override the 275A's current mode and force it to switch to the mode specified by the control signal. When the signal ceases, the 275A returns to the mode that was active before the control signal appeared. (If a front-panel button is pressed while a continuous signal is present, the mode will not change until the signal ends—only then will the 275A return to the newly-selected mode.)

The advantage of using continuous control signals is that the 275A cannot get "hung up" in the wrong mode because the automation failed to notice that one event had ended, and that a new event requiring different 275A processing was on-line. The end of each event is automatically indicated by the end of its control signal. If the "default" mode is AUTO (as it most often will be), then there is no possibility of a catastrophic error, such as loss of audio.

The 275A provides several graceful exits from automation failure. If two or more automation control lines are active simultaneously, the unit will immediately return to AUTO recognition mode. In addition, an AUTOMATION LOCKOUT button (duplicated on the optional Remote Control Panel) allows the operator to lock out all automation signals, and to control the 275A manually until proper automation system operation is restored.

SYNTHESIZER FREQUENCY RESPONSE IN NARROW MODE

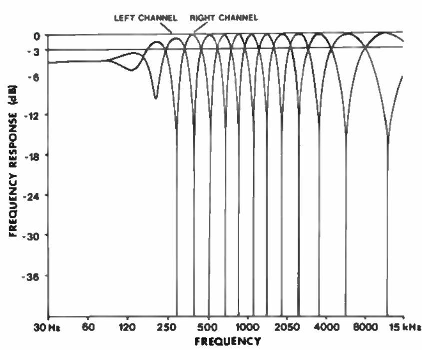


FIGURE 1

Specifications

Frequency Response
(ref mono sum):
± 1/2dB, 30-15,000 Hz (Bypass mode)
± 1dB, 30-15,000 Hz (Synthesize mode)

Total Harmonic Distortion
(+18dBm/600 ohms):
<0.02%, 30-15,000 Hz (Bypass mode)
<0.3%, 30-15,000 Hz (Synthesize mode)

Noise At Output
(30-15,000 Hz):
< −80dBm (Bypass mode)
< −67dBm (Synthesize mode)

Input
Impedance: >10K ohms, balanced bridging.
Absolute overload occurs at +26dBm.

Output
Impedance: <100 ohms, balanced to ground.
Clipping occurs at +26dBm into 600 ohms.

Connectors
Audio: Cinch-type 140-Y barrier strip (#5 screw).
User Control Interface: Type DB-25S jack (accepts DB-25P plug).
275A/RC Remote Control: Type DC-37S jack (accepts DC-37P plug, supplied).

Power Requirements
115-230V AC 50/60Hz, 9VA. Supplied with "U-ground" grounding-type plug to USA standards.

Mounting
Requires 1 unit (1¾", 4.5cm) of vertical space in an EIA 19" (48.3cm) rack.
Depth is 9⅝" (24.5 cm).
Optional 275A/RC remote control unit requires the same space, except depth is 2¼" (5.7cm), including supplied connector.

Shipping Weight
12 lbs. (5.4 kg)

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800041-000-02 2/87

The Orban 275A Automatic Stereo Synthesizer

An unrivaled problem-solver for Stereo TV Transmission

Features

- ☐ **Two methods of automatic mono recognition**
  "Single-Channel" detects absence of audio on one channel, "Mono/Stereo" recognizes mono in both channels.
- ☐ **Two stereo synthesis modes**
  NARROW mode effectively centers dialog, WIDE mode is more dramatic for music and effects.
- ☐ **Smooth cross-fading between true and synthesized stereo**
  No pops, clicks, or discontinuities.
- ☐ **Full mono compatibility**
  Sum of synthesized outputs is identical to mono input.
- ☐ **Patented Orban phase-shift derived comb-filter stereo synthesis technique**
  Provides synthesized stereo without addition of unnatural resonances or "flanging" colorations.
- ☐ **Automatic detection and correction of polarity-reversed ("out-of-phase") stereo inputs**
- ☐ **Noise reduction for mono material**
  Typically 10dB reduction of hiss and hum; single-ended.
- ☐ **Optically-isolated external automation control interface**
- ☐ **Optional remote control unit duplicates main unit functions**
- ☐ **Fully balanced stereo inputs and outputs**
  Can be used with +4dBm or +8dBm lines.

A Stereo Synthesizer That's Right for Stereo TV

The scarcity of true stereo program material has caused many television broadcasters to look for a device that could effectively and automatically reprocess mono material into synthesized stereo. Orban has responded to this need by developing an automatic stereo synthesizer specifically designed for in-line stereo synthesis of mono TV audio.

The **Model 275A Automatic Stereo Synthesizer** improves upon the popular, manually-operated Model 245F Stereo Synthesizer with these added features:

- ☐ Automatic mono/stereo recognition and switching with smooth cross-fade.
- ☐ A choice of *two* methods of automatic mono recognition and *two* stereo synthesis modes.
- ☐ Important stereo television "utility" features like polarity error correction and noise reduction.
- ☐ Exceptionally versatile remote control.

Manual, Automatic, or Remote Control

The 275A Stereo Synthesizer is designed to be placed permanently in the program line. Unless one of its synthesis or utility functions is selected, the 275A will pass audio transparently.

Manual controls on the front panel select whether stereo is to be synthesized from left or right channel mono, whether the "wide" (best-suited for music and effects) or "narrow" (best-suited for voice) synthesis mode is used, and whether noise reduction should be applied to the mono signal prior to stereo synthesis.

Front-panel pushbuttons also allow the operator to activate the automatic recognition and synthesis circuitry, bypass the synthesizer, lock out external automation, or route true-stereo audio through the reversed-polarity detector and corrector.

LED indicators show the functions selected, as well as the operating status of the noise reduction and polarity correction utilities.

All of these controls and indicators are duplicated on the optional, rack-mountable 275A/RC Remote Control Panel.

In addition, the 275A Stereo Synthesizer can be controlled from the station's automation system, tally, or vertical interval decoder through optically-isolated logic inputs available at a rear-panel connector.

Automatic Recognition of Mono/Stereo

It is very difficult to design a circuit which accurately distinguishes between mono and true stereo program material. What defines true mono? If the program material on the left and right channels is identical but differs slightly in level or phase due to recorder or plant errors and tolerances, is that true mono? And should a synthesizer switch-in during those segments of a stereo program when there is no stereo music or sound effects—only center-channel dialog or effects which are electronically identical to true mono?

Although electronic recognition is required, it is clear that no electronic circuit using present-day technology can perform this task perfectly—positive identification of center-channel material still requires the human ability to perceive meaning and high-level context. Nevertheless, at the request of broadcasters Orban is providing such automatic recognition, painstakingly refined and fine-tuned to make fewer errors than other center-channel recognition devices. Because some errors are still inevitable with center-channel recognition, the 275A also provides a more reliable alternative: single-channel recognition.

Single-Channel Recognition: With this approach, the station routes all mono material through the program lines on *one channel only*. If the 275A detects a mute channel, it will automatically synthesize stereo from the other, active channel. If both channels are active, the signal is considered stereo and the synthesizer is bypassed. Single-channel recognition assures reliable detection without recognition errors.

Mono/Stereo Recognition: In this technique, no special routing of program material is required. Audio present on both channels is analyzed by the 275A's recognition circuitry. Even if the program material is primarily hard-center dialog with low-level stereo music or audience noise background, the 275A will recognize it as stereo and bypass the synthesizer. If the material is electrically mono, it is routed to the synthesizer. Minor phase and level differences caused by plant tolerances are ignored. Audio present on only one channel always activates the synthesizer, as in the single-channel technique.

Regardless of which method of recognition you choose, the 275A Stereo Synthesizer cross-fades between synthesized and true stereo material smoothly and unobtrusively.

Stereo Synthesis

The patented Orban stereo synthesis technique creates a compelling pseudo-stereo effect from a mono signal by dividing the audio spectrum into several frequency bands, then directing these bands alternately to the left and right channels. It does this by passing a mono signal through a chain of phase shifters to generate an artificial L-R signal, which is then added to the mono to obtain the synthesized left channel and subtracted from the mono to obtain the synthesized right channel. The net effect is a "complementary comb filter" (see figure 1). The sum of the two synthesized channels always remains equal to the original mono, ensuring mono compatibility.

Because the audio spectrum is divided logarithmically, the undesirable harmonic reinforcement and cancellation which can result from arithmetic band-splitting is avoided. The number of bands (and therefore their individual bandwidths) determines how "dramatic" the stereo effect is. With the 275A Automatic Stereo Synthesizer, two types of remote-selectable stereo synthesis effects are available: a small number of wider bands results in a dramatic sense of stereo space on music and effects (similar to our 245F), while a larger number of narrower bands centers dialogue more accurately. A recessed SEPARATION control adjusts the amount of inter-channel difference (L-R), which determines the relative width of the stereo image.

To maximize loudness while making efficient use of the modestly-priced amplifiers in most consumer TV sets, energy below approximately 200Hz remains mono. The ear can not detect stereo separation in this region.

Noise Reduction

Older mono material often suffers from hiss and other forms of noise. The 275A can apply single-ended noise reduction to mono audio prior to stereo synthesis processing. This noise reduction combines program-controlled high-frequency filtering with broadband expansion. 10dB of noise reduction is typically achieved—without unnaturally reducing ambiance and dialog intelligibility when program levels are low.

Because the noise reduction system is single-ended, no encoding (or later decoding) of the program material is necessary. This makes the process ideally suited to noisy optical soundtracks and satellite feeds. Operation is exceptionally smooth and subtle, and "pumping" and "breathing" are entirely absent.

Noise reduction is not available for true stereo material, since the quality of most stereo material is high and the feature would not justify the additional cost.

Polarity Correction

In stereo material, it is essential that the two channels be in phase with each other. If they are not, the mono sum signal will be seriously degraded as the two channels cancel each other. And that means that the viewer with a mono set (a majority of your audience) will hear disastrously inferior audio, and in some cases no sound at all!

To ensure the mono compatibility of your stereo broadcasts, the 275A can act as a "watchdog" over your program line polarity, correcting errors when detected. The detection technique is very reliable and highly resistant to "falsing"—even when subjected to substantial high-frequency phase errors (due to misaligned heads or other mechanical problems) or when monitoring soundtracks containing out-of-phase "surround" energy.

An LED on the front (and remote control) panel lights when a polarity reversal is being corrected. The detection/correction circuit can be activated or defeated at any time.

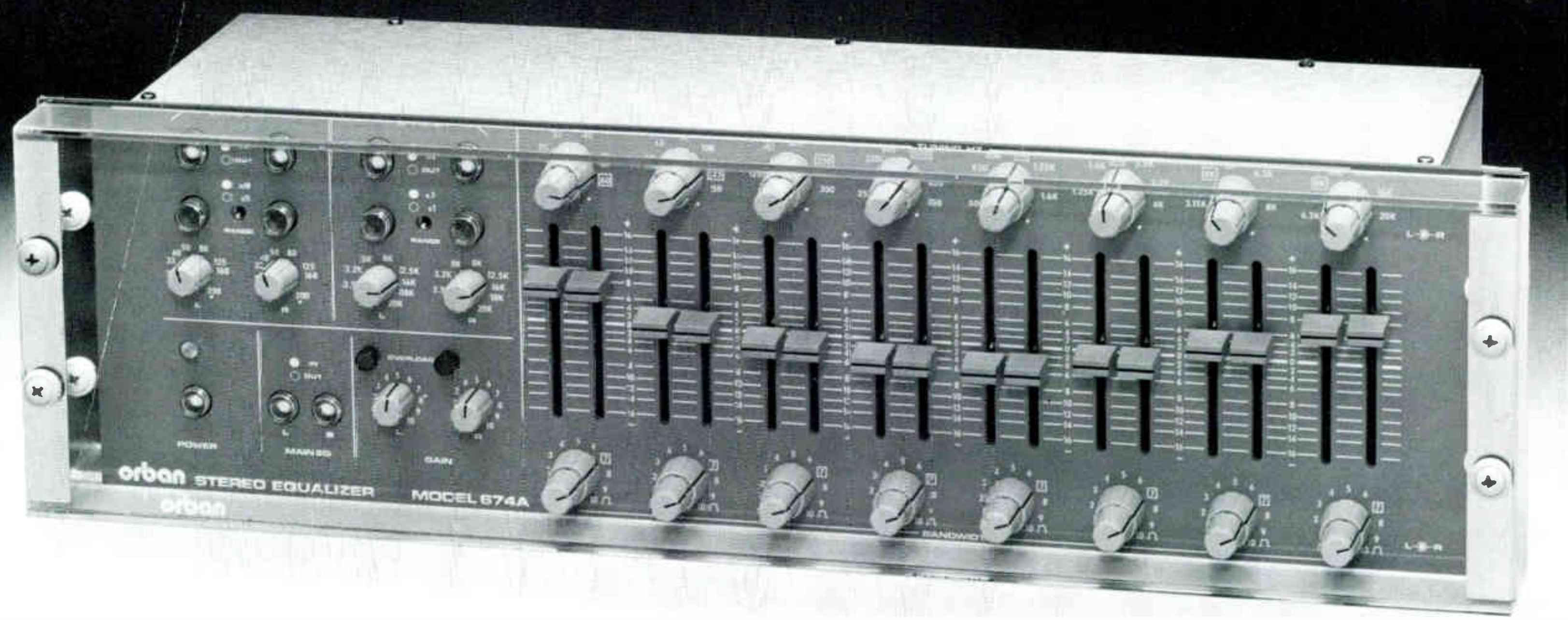
The OPTIMOD Tradition

The 275A Automatic Stereo Synthesizer has been designed to be an integral part of Orban's OPTIMOD® Television Stereo System, along with the:

8182A Audio Processor
8185A/SG Stereo Generator
8182A/SAP Second Audio Program Generator
8182A/PRO Professional Channel Generator

The stability, reliability, and superb performance that distinguish Orban products make them the equipment of choice for the innovative stereo television broadcaster.

Orban Universal Security Cover



This acrylic security cover attaches easily to any Orban product to protect it from fiddling fingers or inquisitive eyes. It fits most other EIA-standard rack-mount panels having a maximum protrusion of 1¼".

It's available in four sizes to fit panels from 1¾" through 7", and in three colors: clear, blue transparent and opaque white. Acrylic may be painted to achieve other effects.

Three sets of screws are supplied: thumbscrew, Phillips head, and hex socket head (with wrench) for three levels of security. Any other 10-32 × ½" screw may be used.

Available through authorized Orban Pro-Audio and Broadcast Dealers

Order Guide:

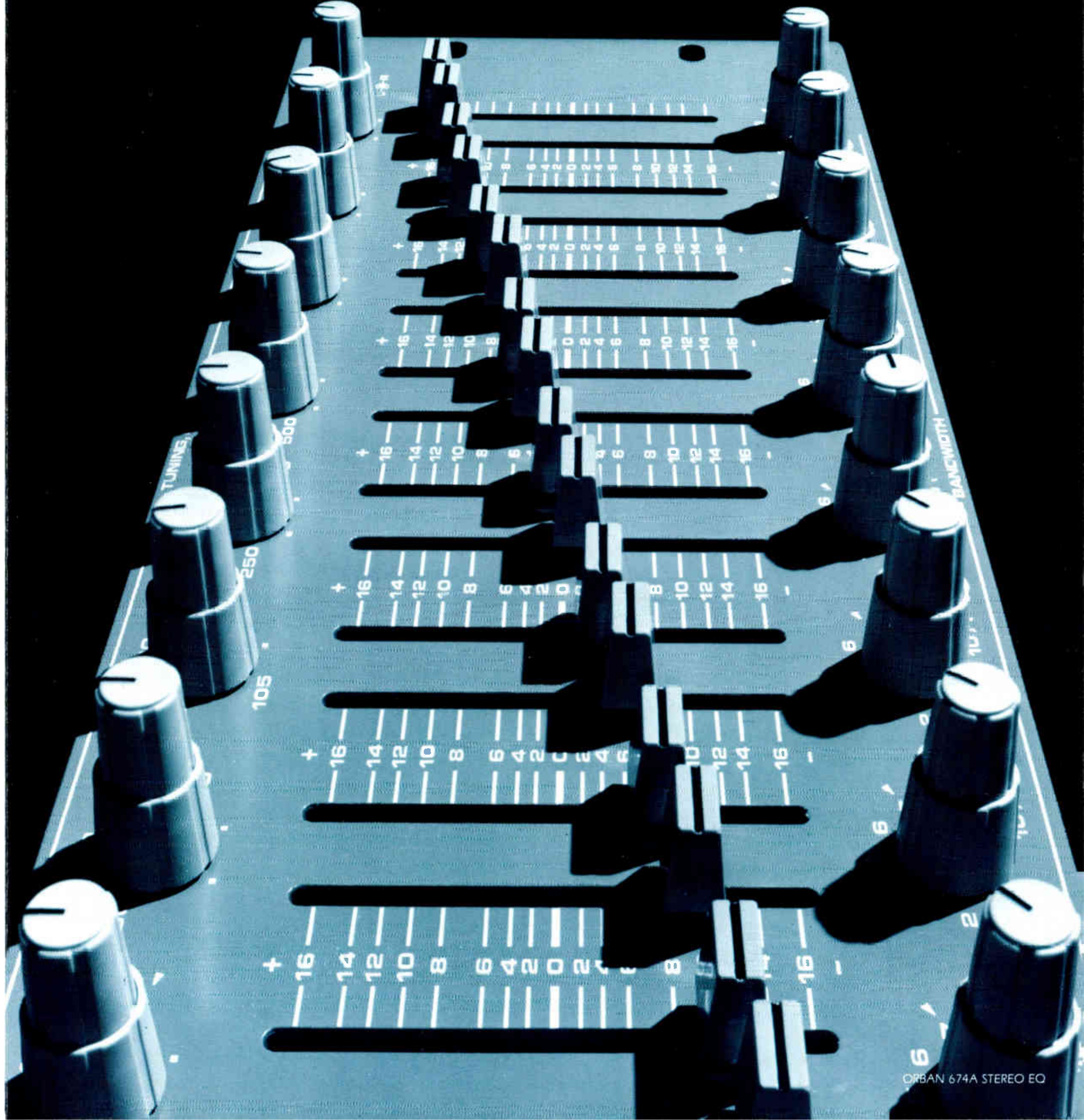
Model	Panel Height	Suffix (xx) for color
ACC-11xx	1¾" (1 unit)	CL Clear Transparent
ACC-12xx	3½" (2 unit)	BL Blue Transparent
ACC-13xx	5¼" (3 unit)	WH Opaque White
ACC-14xx	7" (4 unit)	

orban

Orban Associates Inc., 645 Bryant St., San Francisco, CA 94107 U.S.A. (415) 957-1067 Telex: 17-1480

The Orban 674A Stereo Equalizer

The versatility of a parametric; The economy of a graphic...
More flexible than either.



Technical Description

The 674A Equalizer consists of a balanced input buffer amplifier, eight main equalization amplifiers connected in series, and tunable lowpass and highpass positive feedback 12dB/octave Butterworth filters. The output of the highpass filter is buffered to drive 600 ohms, and is available separately. By suitable switch settings, the main output can be made to carry a lowpassed signal. Thus the 674A can be used as an equalizer cascaded with a full electronic crossover, or as an equalizer cascaded with lowpass and highpass filters.

Each amplifier in the equalizer section provides equalization for one band only, assuring no interaction between bands. The total equalization is simply the sum (in dB) of the equalizations provided by each of the sections.

Peak boost is accomplished by adding the output of a two-pole bandpass resonator to the main signal; reciprocal dip occurs when this resonator is symmetrically connected as a feedback element in the main equalizer amplifier.

The EQ IN/OUT switch bypasses the last seven main amplifiers and defeats equalization in the first amplifier. Gain and signal polarity are equal in the IN and OUT modes. As the BANDWIDTH control is operated, the skirts of the equalization curve move in and out, but the peak gain and peak frequency remain constant. As the EQUALIZATION controls are operated, the frequencies of peak gain remain constant. However, as the TUNING control or EQUALIZATION control (in **dip** mode) are operated, the bandwidth ("Q") will change, because of the simplifications in the "quasi-parametric" bandpass resonator. Careful design has enabled us to produce curves (in **boost** mode only) essentially identical to the desirable "constant-Q" curves provided by our 622B **true parametric** equalizer in its **boost** mode.

The EQUALIZATION controls all produce peaking curves; if shelving curves are desired, they can be approximated by tuning the lowest band to 20Hz and the highest band to 20kHz. The breakpoint of the shelving characteristic is then adjusted with the BANDWIDTH control.

Summary

Many people are now aware of the **power** of parametric equalization: the almost sensual satisfaction of getting the sound really **right.** These same people are also demanding professionals, insisting on inaudible noise and distortion, human engineering, quality "feel", and uncompromising reliability.

Orban is well-known for its line of fine parametric equalizers, like the 622B. Now with the 674A, it brings equalization of the same rigorous quality to applications where it could never before be afforded. The 674A is inexpensive enough to qualify it for serious consideration in applications which would otherwise be given by default to a much less able graphic equalizer.

The 674A rounds out the line of Orban "Professionals' Parametrics." Between the 622B **true parametric** and the 674A **quasi-parametric,** there is an equalizer for virtually every need and budget. The Orban "Professionals' Parametrics" are available at your authorized Orban dealer.

Specifications:

All specifications apply when driving 600 ohms or higher impedances. Noise measured on an average-reading meter through a 20-20,000Hz bandpass filter with 18dB/octave Butterworth skirts.

ELECTRICAL

Input:
Impedance, Load (each leg): 100K in parallel with 1000pF, electronically balanced
Impedance, Driving: Ideally 600 ohms or less, balanced or unbalanced
Nominal Input Level: Between −10 and +4dBm
Absolute Overload Point: +26dBm

Output:
Impedance, Source: 47 ohms in parallel with 1000pF, unbalanced (Optional transformer balanced 600 ohm outputs)
Impedance, Load: Should be 600 ohms or greater—will not ring into any capacitive load
Nominal Output Level: +4dBm
Max. Output Level Before Clipping: greater than +19dBm, 20-20,000Hz

Frequency Response:
±0.25dB; 20-20,000Hz: EQ controls set at zero detents

Available Gain:
+12dB; adjustable to -infinity by means of front-panel GAIN control

Slew Rate:
Varies between 6 and 13V/us depending upon setting of GAIN controls; slewing is symmetrical. Internal bandlimiting assures that slew rate limiting will not occur even with the most severe equalization and program material.

Square Wave Response:
Square wave exhibits no spurious ringing at any output level. The only ringing observable is that theoretically associated with any given equalization curve.

Total Harmonic Distortion:
Less than 0.08%, 20-20,000Hz (+18dBm)

SMPTE Intermodulation Distortion:
Less than 0.05% (+18dBm: 60/7000Hz, 4:1)

Noise at Output:
Less than −78dBm (EQ in, filters out, controls centered)

Overload/Noise Ratio:
Better than 113dB for any single bandpass filter, for any settings of TUNING or BANDWIDTH controls.

Equalization Ranges:
±16dB peaking EQ, Reciprocal

Tuning Ranges:
20-60Hz; 40-150Hz; 110-310Hz; 230-750Hz; 480-1900Hz; 1.1-4.5Hz; 2.8-9.0kHz; 5.9-21kHz. Dials calibrated at ISO preferred frequencies.

Crosstalk:
Typically better than −55dB @ 20kHz; improves at 6dB/octave below that frequency.

"Q" Range:
Greater than 0.5 to 10 for any setting of the TUNING control.

Low Pass Filter Section:
Tunable in 2 ranges: 200-2000Hz or 2.0-20kHz, 12dB/octave, (2nd-order Butterworth)

High Pass Filter Section:
Tunable in 2 ranges: 20-200Hz or 200-2000Hz, 12dB/octave, (2nd-order Butterworth)

Overload Indicator:
Lamp lights for 200ms if the instantaneous peak output of any amplifier rises to within 1dB of its clipping point.

Circuit Design:
Active RC realized with FET-input opamps. Line driver employs discrete transistor current booster.

Operating Temperature:
0-50° C

Power Requirements:
115/230VAC ±10%; 50/60Hz; 12 watts

PHYSICAL

Operating Controls (each channel):
EQUALIZATION, TUNING, and BANDWIDTH for each of eight bands. TUNING, RANGE (×1; ×10), and FILTER IN/OUT for each filter. EQUALIZATION IN/OUT, POWER ON/OFF, and GAIN for entire equalizer.

Panel:
19" × 5¼" (48.3 × 13.3cm): 3 units

Chassis Depth Behind Panel:
5¼" (13.3cm)

Weight:
Net: 11lbs. (5 kg); Shipping: 13½ lbs. (6.1 kg)

AC Cord:
3-wire U-ground to USA Standard

Connectors:
140 type barrier strip (5# screw); holes punched for XLR-type connectors (Switchcraft D3F and D3M or equal)

Circuit Ground:
Available on barrier strip; normally jumpered to chassis.

Options:
1) Plexiglass security cover for EQ and filter sections
2) Balanced transformers in two or four outputs
3) XLR-type connectors on input and two or four outputs
4) Phone jacks on input and two or four outputs

Specifications subject to change without notice

PERFORMANCE HIGHLIGHTS

EQ Section

- ☐ Eight bands per channel, each with TUNING AND BANDWIDTH controls
- ☐ Each band tunes over 3:1 frequency range
- ☐ "Q" typically variable between 0.3 and 20 (center TUNING)
- ☐ ±16dB equalization range
- ☐ EQ controls are long-throw dust-shielded slidepots for good resolution
- ☐ TUNING and BANDWIDTH controls marked with "tics" indicating typical settings
- ☐ Narrowband notching capability ideal for sound reinforcement
- ☐ Bands totally non-interacting
- ☐ A dedicated stereo device with controls arranged for optimum ease of maintaining stereo balance

HP/LP Filter Sections

- ☐ Each section continuously tunable over 100:1 range in 2 decades
- ☐ Each section independently switchable
- ☐ 12 dB/octave slopes
- ☐ Filters follow graphic section. Separate main/lowpass and highpass outputs allow use as filters or as full electronic crossover

General

- ☐ Very low noise and distortion
- ☐ High slew rate for minimum TIM (SID)
- ☐ Front-panel GAIN controls; 12dB gain available
- ☐ "Peak-stretching" overload lamps warn of clipping anywhere in equalizer
- ☐ Active balanced inputs; unbalanced outputs. Transformer-balanced outputs optional
- ☐ RF suppression on inputs, outputs, and power leads
- ☐ 115/230V, 50-60Hz transformer is standard
- ☐ Industrial-grade parts and construction including socketed IC's
- ☐ Highly cost-effective

Optional Security Cover (ACC-3) available for filter/crossover section only in addition to ACC-13 which covers entire unit.

The 674A Stereo Equalizer

The Orban 674A is a cost-effective, professional, quasi-parametric equalizer with the convenience of graphic-type EQ controls. Wide-range high- and low-pass filters with 12dB/octave Butterworth slopes follow the EQ section for added versatility. The 674A has two outputs per channel, arranged so that these filters can also be used as a fully tunable electronic crossover.

The space-saving 674A offers the facilities of two complete mono 672A's in a single chassis. Ganged, concentric controls make one-hand stereo operation of bandwidth and tuning a snap. Graphic-style EQ controls are split parallel for each of the eight bands. Separate high-and low-pass filters on each channel offer stereo two-way electronic crossover capability.

While it is possible to operate the unit with two unrelated mono program sources, under some circumstances high-frequency crosstalk may be experienced. Therefore, crosstalk requirements should be evaluated in such applications. (See Specifications)

The 674A is a professional product designed to provide a large measure of versatility, convenience, and quality at a very attractive price. While it meets the requirements of the demanding professional, it is also designed and priced to make it understandable and available to the advanced audiophile.

To make the 674A easy to use in situations where its full versatility isn't needed, "tic" marks have been included on the dial calibrations of the TUNING and BANDWIDTH controls. When these controls are set to the tics, the 674A behaves like a standard octave-band graphic equalizer with the eight bands on ISO frequencies from 63 to 8000Hz.

Each feature of the 674A has been thoughtfully chosen and cleverly implemented to make the equalizer a particularly powerful tool in nearly all areas of audio: sound reinforcement, public address, recording studio, broadcasting, motion picture sound, disco, theater . . .

Why "Quasi-Parametric"?

There are two basic types of parametric equalizer: **full-** and **quasi-**parametric. Orban manufactures both types. Both offer far more effective control than other kinds of equalizers, like graphics. Our popular dual-channel 622B is a **full** parametric. This means that you have **totally non-interacting** control over the three fundamental **parameters** of equalization: the **amount** of peak boost or dip (in dB), the **tuning** (the frequency most affected by the equalization), and the "Q" (which relates to the sharpness of the EQ curve—the degree to which frequencies on either side of the peak frequency are affected by the equalization). As opposed to our 622B Equalizer, the 674A is **quasi**-parametric. This means that the "Q" changes when you adjust the TUNING and/or EQ controls. Other control adjustments are completely non-interacting: TUNING and EQ do not affect each other.

The other important performance difference between the full-parametric 622B and the new 674A is that the 622B's EQ curves are **"constant-Q"**; the 674A's curves are "reciprocal". "Constant Q" curves are valuable in that they permit infinite-depth notches to be created; reciprocal curves limit the maximum cut to the same number of dB as the maximum boost. In the case of the 674A, 16dB of cut is available. This is fine for tuning out ring-modes in sound reinforcement systems, but might not be adequate in all circumstances to remove hum or other fixed-frequency interference from a signal. On the other hand, some people prefer reciprocal curves because the boost and cut are mirror images of each other, thus permitting previous equalization to be readily "undone" later. Careful design of the circuitry gives the 674A **in boost mode** a characteristic similar to the 622B's desirable "constant-Q" curve family.

Why did we choose the quasi-parametric technique for the 674A? Because it offers a way to produce a very high quality, stable equalizer at low cost without compromising distortion, noise, accuracy, or reliability.

Applications

Sound Reinforcement and Monitor Tuning

There are many ways to use the 674A in sound reinforcement and monitor tuning:

1) In an economy biamped installation, replace both the third-octave equalizer **and** the electronic crossover with the 674A. The 674A's narrowband, tunable notches can deal with ring-modes more effectively than the third-octave unit could. Use three or four of the 674A bands for narrowband notching; leave the rest for wideband EQ.

2) In a higher budget biamped installation, use the 674A as an electronic crossover plus a narrowband, tunable notch filter for ring-mode suppression; incorporate a separate third-octave equalizer to correct the house curve.

3) Use variations of (1) and (2) above **with** an electronic crossover; the 674A's highpass and lowpass filters can then be used to roll off the frequency response of the system in a controlled manner.

4) In a **non-biamped** system (like a stage monitor), use the 674A to equalize the monitor, and use its filters to restrict response in the extreme high and low frequencies.

5) Use the 674A as a **partial** electronic crossover plus equalizer/filter by devoting one channel of 674A equalization to each driver; one filter is required to perform the crossover function; the other can be used for its normal highpass (or lowpass) function.

6) For super power in mono reinforcement applications, connect both channels in series. You'll then get **sixteen** EQ/notch filter bands, an electronic crossover, and an extra set of filters to limit system bandwidth.

In all cases, the BANDWIDTH control can be adjusted to make the totally non-interacting (series-connected) bands "combine"—a most desirable characteristic in sound reinforcement.

Any way you cut it, the 674A's economy and extraordinary versatility make it one of the sound reinforcement practitioners' most useful tools.

Recording Studios

Every recording studio needs a few channels of 674A equalization to handle the tough chores that the internal console equalizers can't deal with. Patch that problem track through a 674A: its fine-tuning ability lets you clean up the track far more effectively than you could with a graphic or "three knob" console equalizer. Use the tunable filters to help eliminate rumble, cymbal splash, kick drum leakage—you name it!

If you need to correct the equalization of a track because of second thoughts during or after the mix, the 674A can create the finishing touches as no ordinary equalizer can. It's better than a third-octave graphic, because the 674A can generate broad, non-ringing boosts, whereas the graphic is much more colored and ringy.

The 674A is also an ideal adjunct to an electronic music synthesizer—you can create high "Q" formants and shape the spectrum so that the sound comes alive.

Motion Picture Sound

The 674A is an ideal replacement for the graphic equalizers ordinarily used for dialogue equalization in motion picture sound. Set the TUNING and BANDWIDTH controls to the "tics" on the panel, and you get the equivalent of a familiar easy-to-operate graphic. But when you **need** the extra control and flexibility—such as notching out the extraneous sounds that always seem to plague location recordings—that power is there instantly, without patching or the use of external dip filters. The high- and lowpass filters are invaluable for cleaning up noise and rumble without affecting dialogue—and without using up EQ channels to try to achieve filtering. In addition, many "effects" (such as telephone, pocket radio, or "old time" recordings) can be easily created with the 674A alone.

In addition, the 674A can be used to equalize the "B-chain" in the re-recording theater to the acoustic response specifications of the studio. The lowpass filter can effectively simulate the "Academy Rolloff" or its current modifications.

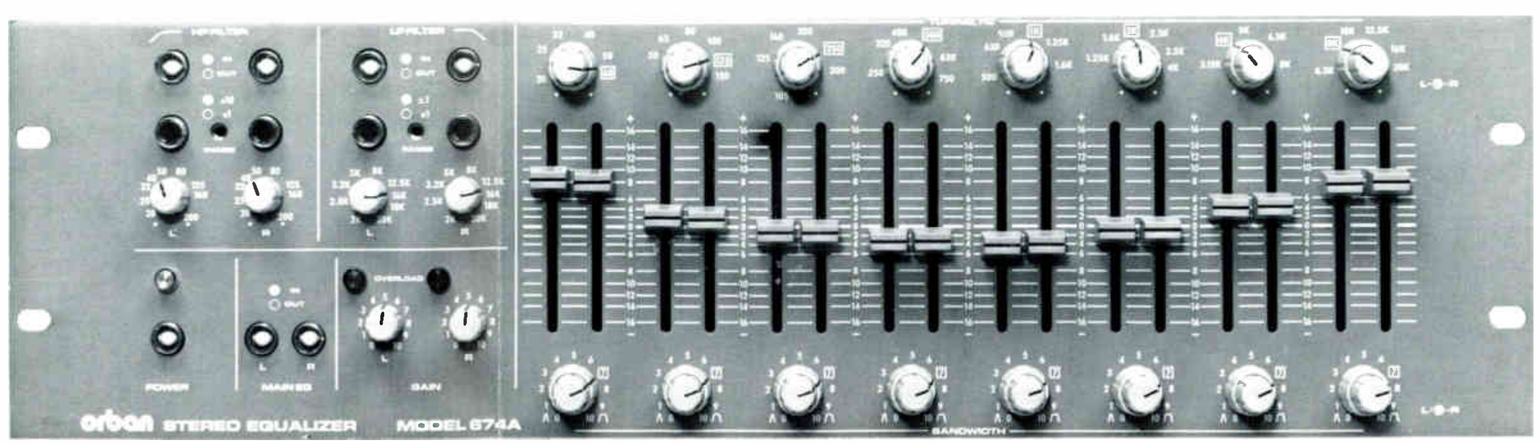
Stereo Broadcasting

Use the 674A in the production studio to enhance the announce mike, "sweeten" stereo music, and to create special production effects that make your station stand out among its competitors. Meanwhile, another 674A can be quietly and efficiently equalizing the stereo program line for maximum punch and brightness on the air. Use the 674A to equalize phone or remote lines for flat response—it's much more versatile than the standard phone company equalizers. In the main studio, use it on the announce mike channel to equalize for maximum presence, and also to notch out sounds like mechanical hum from cart machine motors or air conditioning noise. Whatever your application, the 674A's RF suppression and optional output transformer mean problem-free installation in high-RF environments.

Dance Bars

The 674A is an excellent dance bar stereo equalizer. The sound contractor installing the system can offer the management exactly the sound desired—including solid, punchy bass free from muddiness and boom—and an aggressive, sizzling top free from ringing and coloration typical of a full-octave graphic equalizer. The eight bands permit substantial work to be done in flattening out undesired response deviations in the upper bass and midrange. Narrowband notches can even deal with the difficult resonances sometimes encountered in high-efficiency horn-type loudspeakers. In biamped installations, use the separate lowpass and highpass filter outputs as a complete electronic crossover. No other crossover is necessary.

The 674A costs a bit more than an octave-type graphic. But, unlike a graphic, it really **solves** the problem.



orban

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Telex: 17-1480
Cable: ORBANAUDIO

Technical Description

The 672A Equalizer consists of a balanced input buffer amplifier, eight main equalization amplifiers connected in series, and tunable lowpass and highpass positive feedback 12dB/octave Butterworth filters. The output of the lowpass filter is buffered to drive 600 ohms, and is available separately. By suitable switch settings, the main output can be made to carry a high-passed signal. Thus the 672A can be used as an equalizer cascaded with a full electronic crossover, or as an equalizer cascaded with lowpass and highpass filters.

Each amplifier in the equalizer section provides equalization for one band only, thus assuring no interaction between bands. The total equalization is simply the sum (in dB) of the equalizations provided by each of the sections.

Peak boost is accomplished by adding the output of a two-pole bandpass resonator to the main signal; reciprocal dip occurs when this resonator is symmetrically connected as a feedback element in the main equalizer amplifier.

The EQ IN/OUT switch bypasses the last seven main amplifiers and defeats equalization in the first amplifier. Gain and signal polarity are equal in the IN and OUT modes. As the BANDWIDTH control is operated, the skirts of the equalization curve move in and out, but the peak gain and peak frequency remain constant. As the EQUALIZATION controls are operated, the frequencies of peak gain remain constant. However, as the TUNING control or EQUALIZATION control (in **dip** mode) are operated, the bandwidth ("Q") will change, because of the simplifications in the "quasi-parametric" bandpass resonator. Careful design has enabled us to produce curves (in **boost** mode only) essentially identical to the desirable "constant-Q" curves provided by our 622B **true parametric** equalizer in its **boost** mode.

The EQUALIZATION controls all produce peaking curves; if shelving curves are desired, they can be approximated by tuning the lowest band to 20Hz and the highest band to 20kHz. The breakpoint of the shelving characteristic is then adjusted with the BANDWIDTH control.

Summary

Many people are now aware of the **power** of parametric equalization: the almost sensual satisfaction of getting the sound really **right.** These same people are also demanding professionals, insisting on inaudible noise and distortion, human engineering, quality "feel", and uncompromising reliability.

Orban is well-known for its line of fine parametric equalizers, like the 622B. Now with the 672A, it brings equalization of the same rigorous quality to applications where it could never before be afforded. The 672A is inexpensive enough to qualify it for serious consideration in applications which would otherwise be given by default to a much less able graphic equalizer.

The 672A rounds out the line of Orban "Professionals' Parametrics." Between the 622B **true parametric** and the 672A **quasi-parametric,** there is an equalizer for virtually every need and budget. The Orban "Professionals' Parametrics" are available at your authorized Orban dealer.

Specifications

All specifications apply when driving 600 ohms or higher impedances. Noise measured on an average-reading meter through a 20-20,000Hz bandpass filter with 18dB/octave Butterworth skirts.

ELECTRICAL

Input:
Impedance, Load (each leg): 100K in parallel with 1000pF, electronically balanced Impedance, Driving: Ideally 600 ohms or less, balanced or unbalanced.
Nominal Input Level: Between −10 and +4dBm
Absolute Overload Point: +26dBm

Output:
Impedance, Source: 47 ohms in parallel with 1000pF, unbalanced (Optional transformer balanced 6000 ohm outputs)
Impedance, Load: Should be 600 ohms or greater—will not ring into any capacitive load
Nominal Output Level: +4dBm
Max. Output Level Before Clipping: greater than ±19 dBm, 20-20,000Hz

Frequency Response:
±0.25dB, 20-20,000Hz: EQ controls set at "0" detents

Available Gain:
+12dB; adjustable to −infinity by means of front-panel GAIN control

Slew Rate:
Varies between 6 and 13V/uS depending upon setting of GAIN control; slewing is symmetrical. Internal bandlimiting assures that slew rate limiting will not occur even with the most severe equalization and program material.

Square Wave Response:
Square wave exhibits no spurious ringing at any output level. The only ringing observable is that theoretically associated with any given equalization curve.

Total Harmonic Distortion:
Less than 0.05%, 20-20,000Hz (+18dBm)

SMPTE Intermodulation Distortion:
Less than 0.05% (+18dBm:60/7000Hz, 4:1)

Noise at Output:
Less than −75dBm (EQ in, filters out, controls centered)

Overload/Noise Ratio:
Better than 113dB for any single bandpass filter, for any settings of TUNING or BANDWIDTH controls.

Equalization Ranges:
±16dB peaking EQ, Reciprocal

Tuning Ranges:
20-60Hz; 40-150Hz; 110-310Hz; 230-750Hz; 480-1900Hz; 1.1-4.5Hz; 2.8-9.0kHz; 5.9-21.kHz. Dials calibrated at ISO preferred frequencies.

"Q" Range:
Greater than 0.5 to 10 for any setting of the TUNING control

Low Pass Filter Section:
Tunable in 2 ranges: 200-2000Hz or 2.0-20kHz, 12dB/octave, (2nd-order Butterworth)

High Pass Filter Section:
Tunable in 2 ranges: 20-200Hz or 200-2000Hz, 12dB/octave, (2nd-order Butterworth)

Overload Indicator:
Lamp lights for 200mS if the instantaneous peak output of any amplifier rises to within 1dB of its clipping point.

Circuit Design:
Active RC realized with FET-input opamps. Line driver employs discrete transistor current booster.

Operating Temperature:
0-50°C

Power Requirements
115/230VAC±10%; 50/60Hz; 6 watts

PHYSICAL

Operating Controls
EQUALIZATION, TUNING, and BANDWIDTH for each of eight bands. TUNING, RANGE (x1; x10), and FILTER IN/OUT for each filter. EQUALIZATION IN/OUT, POWER ON/OFF, and GAIN for entire equalizer.

Panel:
19" x 5¼" (48.3 x 13.3cm): 3 units

Chassis Depth Behind Panel:
5¼" (13.3cm)

Weight:
Net: 8 lbs. (3.6kg); Shipping: 12 lbs (5.4kg)

AC Cord:
3-wire U-ground to USA Standard

Connectors:
140 type barrier strip (#5 screw) plus parallel-wired ¼" 3 ckt. phone jacks (Switchcraft 12B or equal). Holes punched for XLR-type connectors (Switchcraft D3F and D3M or equal)

Circuit Ground:
Available on barrier strip; normally jumpered to chassis.

Specifications subject to change without notice.

orban

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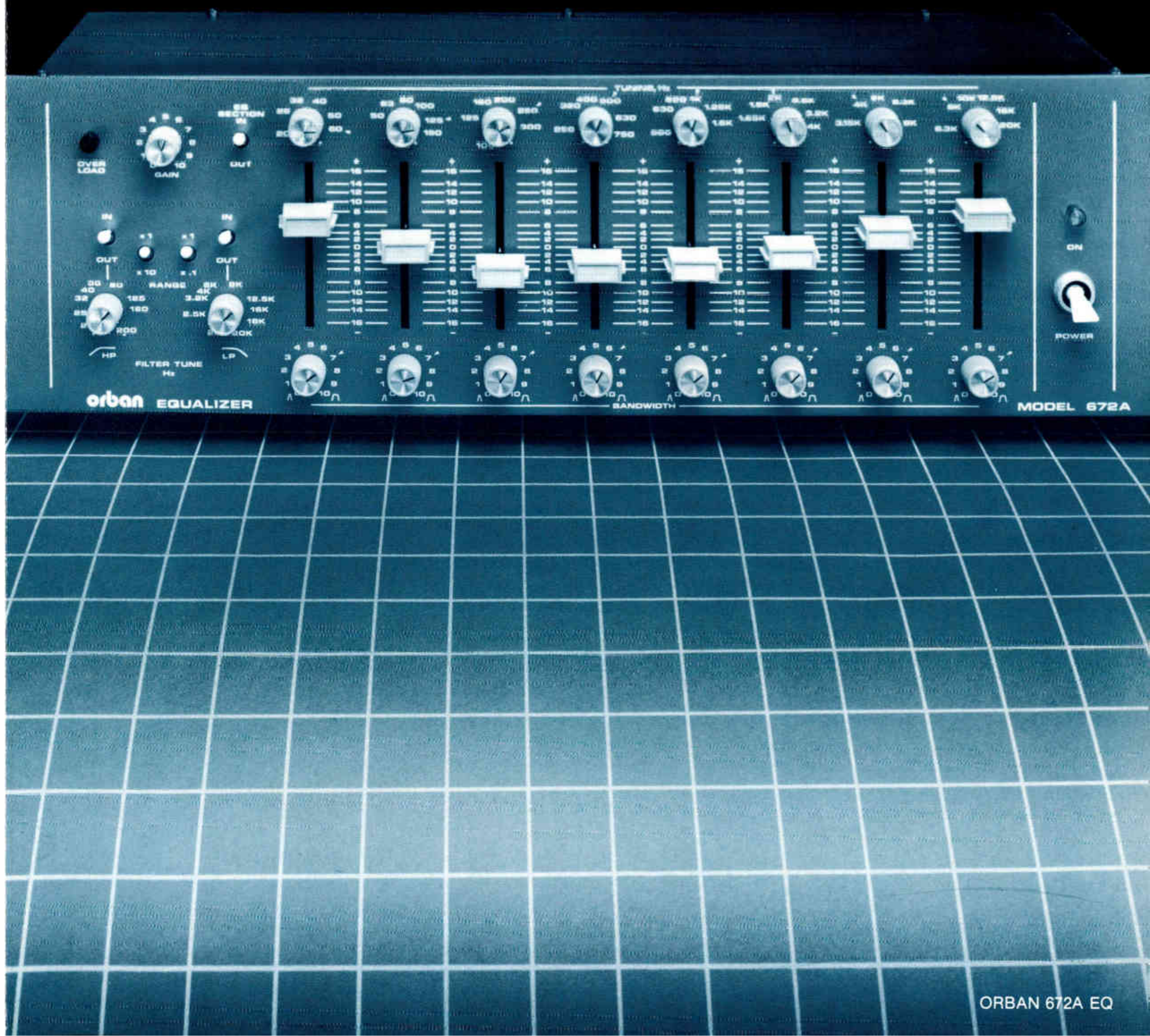
The Orban 672A Equalizer

The versatility of a parametric; The economy of a graphic...
More flexible than either.



PERFORMANCE HIGHLIGHTS

EQ Section

- ☐ Eight bands, each with TUNING and BANDWIDTH control
- ☐ Each band tunes over 3:1 frequency range
- ☐ "Q" typically variable between 0.3 and 20 (center TUNING)
- ☐ ± 16dB equalization range
- ☐ EQ controls are long-throw dust-shielded slidepots for good resolution
- ☐ TUNING and BANDWIDTH controls marked with "tics" indicating typical settings
- ☐ Narrowband notching capability ideal for sound reinforcement
- ☐ Bands totally non-interacting

HP/LP Filter Sections

- ☐ Each section continuously tunable over 100:1 range in 2 decades
- ☐ Each section independently switchable
- ☐ 12dB/octave slopes
- ☐ Filters follow graphic section. Separate main/highpass and lowpass outputs allow use as filters or as full electronic crossover

General

- ☐ Very low noise and distortion
- ☐ High slew rate for minimum TIM (SID)
- ☐ Front-panel GAIN control; 12dB gain available
- ☐ "Peak-stretching" overload lamp warns of clipping anywhere in equalizer
- ☐ Active balanced input; unbalanced outputs. Transformer-balanced outputs optional
- ☐ RF suppression on input, output, and power leads
- ☐ 115/230V, 50-60Hz transformer is standard
- ☐ Industrial-grade parts and construction including socketed IC's
- ☐ Highly cost-effective

A stereo version of this product is also available as Model 674A

INTRODUCING THE 672A EQUALIZER

The Orban 672A is a cost-effective, professional, quasi-parametric equalizer with the convenience of graphic-type EQ controls. Wide-range high- and low-pass filters with 12dB/octave Butterworth slopes follow the EQ section for added versatility. The 672A has two outputs, arranged so that these filters can also be used as a fully tunable electronic crossover.

The 672A is a professional product designed to provide a large measure of versatility, convenience, and quality at a very attractive price. While it meets the requirements of the demanding professional, it is also designed and priced to make it understandable and available to the advanced audiophile.

To make the 672A easy to use in situations where its full versatility isn't needed, "tic" marks have been included on the dial calibrations of the TUNING and BANDWIDTH controls. When these controls are set to the tics, the 672A behaves like a standard octave-band graphic equalizer with the eight bands on ISO frequencies from 63 to 8000Hz.

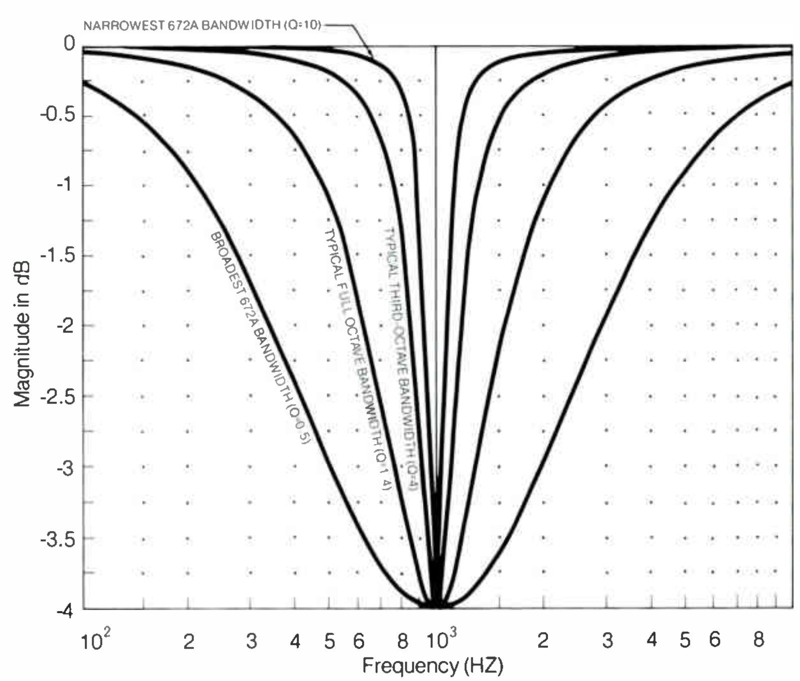
Each feature of the 672A has been thoughtfully chosen and cleverly implemented to make the equalizer a particularly powerful tool in nearly all areas of audio: sound reinforcement, public address, recording studio, broadcasting, motion picture sound, disco, theater . . .

Why "Quasi-Parametric"?

There are two basic types of parametric equalizer: **full-** and **quasi-** parametric. Orban manufactures both types. Both offer far more effective control than other kinds of equalizers, like graphics. Our popular dual-channel 622B is a **full** parametric. This means that you have **totally non-interacting** control over the three fundamental **parameters** of equalization: the **amount** of peak boost or dip (in dB), the **tuning** (the frequency most affected by the equalization), and the **"Q"** (which relates to the sharpness of the EQ curve—the degree to which frequencies on either side of the peak frequency are affected by the equalization). As opposed to our 622B Equalizer, the 672A is **quasi-**parametric. This means that the "Q" changes when you adjust the TUNING and/or EQ controls. Other control adjustments are completely non-interacting: TUNING and EQ do not affect each other.

The other important performance difference between the full-parametric 622B and the new 672A is that the 622B's EQ curves are "**constant-Q**"; the 672A's curves are "reciprocal." "Constant-Q" curves are valuable in that they permit infinite-depth notches to be created; reciprocal curves limit the maximum cut to the same number of dB as the maximum boost. In the case of the 672A, 16dB of cut is available. This is fine for tuning out ring-modes in sound reinforcement systems, but might not be adequate in all circumstances to remove hum or other fixed-frequency interference from a signal. On the other hand, some people prefer reciprocal curves because the boost and cut are mirror images of each other, thus permitting previous equalization to be readily "undone" later. Careful design of the circuitry gives the 672A **in boost mode** a characteristic similar to the 622B's desirable "constant-Q" curve family.

Why did we choose the quasi-parametric technique for the 672A? Because it offers a way to produce a very high quality, stable equalizer at low cost without compromising distortion, noise, accuracy, or reliability.



Various Bandwidth Dipping Curves — 4dB Dip

APPLICATIONS

Sound Reinforcement and Monitor Tuning

There are many ways to use the 672A in sound reinforcement:

1) In an economy biamped installation, replace both the third-octave equalizer **and** the electronic crossover with the 672A. The 672A's narrowband, tunable notches can deal with ring modes more effectively than the third-octave unit could. Use three or four of the 672A bands for narrowband notching; leave the rest for wideband EQ.
2. In a higher budget biamped installation, use the 672A as an electronic crossover plus a narrowband, tunable notch filter for ring mode suppression; incorporate a separate third-octave equalizer to correct the house curve.
3. Use variations of (1) and (2) above **with** an electronic crossover; the 672A's high-pass and lowpass filters can then be used to roll off the frequency response of the system in a controlled manner.
4) In a **non-biamped** system (like a stage monitor), use the 672A to equalize the monitor, and use its filters to restrict response in the extreme high and low frequencies.
5) Use the 672A as a **partial** electronic crossover plus equalizer/filter by devoting one channel of 672A equalization to each driver; one filter is required to perform the crossover function; the other can be used for its normal highpass (or lowpass) function.

In all cases, the BANDWIDTH control can be adjusted to make the totally non-interacting (series-connected) bands "combine" —a most desirable characteristic in sound reinforcement.

Any way you cut it, the 672A's economy and extraordinary versatility make it one of the sound reinforcement practitioner's most useful tools.

Recording Studios

Every recording studio needs a few channels of 672A equalization to handle the tough chores that the internal console equalizers can't deal with. Patch that problem track through a 672A: its fine-tuning ability lets you clean up the track far more effectively than you could with a graphic or "three knob" console equalizer. Use the tunable filters to help eliminate rumble, cymbal splash, kick drum leakage—you name it!

The 672A is also an ideal adjunct to an electronic music synthesizer—you can create high "Q" formants and shape the spectrum so that the sound comes alive.

If you need to correct the equalization of a finished track because of second thoughts after the mix, the 672A can create the finishing touches as no ordinary equalizer can. It's better than a third-octave graphic, because the 672A can generate broad, non-ringing boosts, whereas the graphic is much more colored and ringy.

Motion Picture Sound and Video Sweetening

The 672A is an ideal replacement for the graphic equalizers ordinarily used for dialogue equalization in motion picture sound. Set the TUNING and BANDWIDTH controls to the "tics" on the panel, and you get the equivalent of a familiar, easy-to-operate graphic. But when you **need** the extra control and flexibility—such as notching out the extraneous sounds that always seem to plague location recordings—that power is there instantly, without patching or the use of external dip filters. The high and lowpass filters are invaluable for cleaning up noise and rumble without affecting dialogue—and without using up EQ channels to try to achieve filtering. In addition, many "effects" (such as telephone, pocket radio, or "old time" recordings) can be easily created with the 672A alone.

In addition, the 672A can be used to equalize the "B-chain" in the re-recording theater to the acoustic response specifications of the studio. The lowpass filter can effectively simulate the "Academy Rolloff" or its current modifications.

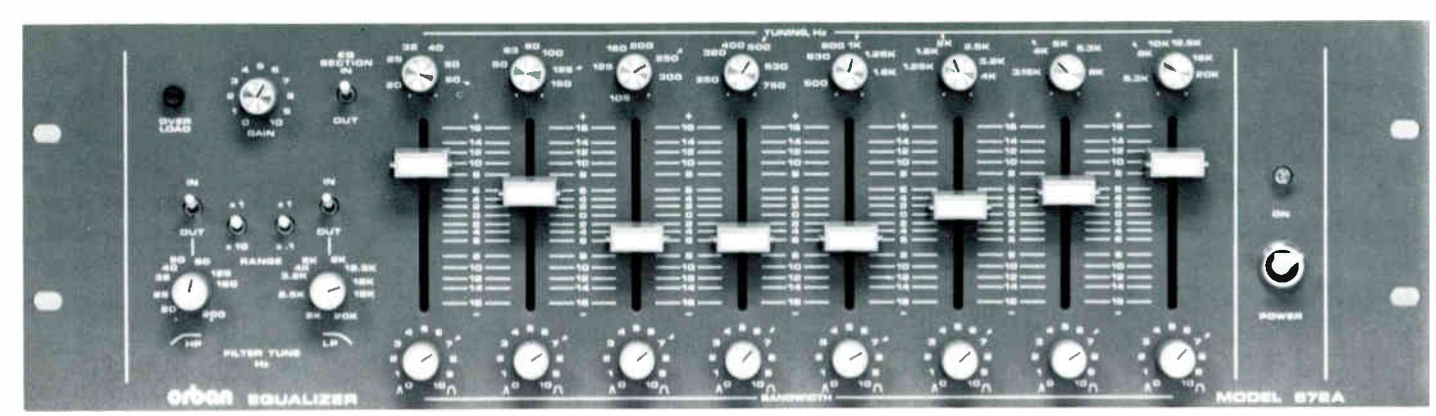
Broadcasting

Use the 672A in the production studio to enhance the announce mike, and to create special production effects that make your station stand out among its competitors. Meanwhile, another 672A can be quietly and efficiently equalizing the program line for maximum punch and brightness on the air. Use the 672A to equalize phone or remote lines for flat response—it's much more versatile than the standard phone company equalizers. In the main studio, use it on the announce mike channel to equalize for maximum presence, and also to notch out sounds like mechanical hum from cart machine motors or air conditioning noise. Whatever your application, the 672A's RF suppression and optional output transformer mean problem-free installation in high-RF environments.

Dance Bars

The 672A is an excellent dance bar equalizer. The sound contractor installing the system can offer the management exactly the sound desired—including solid, punchy bass free from muddiness and boom—and an aggressive, sizzling top free from the ringing and coloration typical of a full-octave graphic equalizer. The eight bands permit substantial work to be done in flattening out undesired response deviations in the upper bass and midrange. Narrowband notches can even deal with the difficult resonances sometimes encountered in high-efficiency horn-type loudspeakers. In biamped installations, use the separate lowpass and highpass filter outputs as a complete electronic crossover. No other crossover is necessary.

The 672A costs a bit more than an octave-type graphic. But, unlike a graphic, it really **solves** the problems.



Technical Description

The 622 Parametric Equalizer consists of a balanced input buffer amplifier, an input attenuator, and four peak/dip equalization sections connected in series, assuring no interaction between sections. The final section contains a current booster capable of driving 600 ohm loads. The output of the input buffer and each of the equalization sections is monitored at all times by the overload indicator. The EQ IN/OUT switch bypasses all circuitry but the input buffer and output amplifier; it is arranged so that gain and signal polarity are maintained constant in the IN and OUT modes.

Equalization is accomplished by summing the output of a two-pole bandpass filter to the main signal in-phase (for boost) or out-of-phase (for cut). This creates the "Constant-Q" curves described above.

As the BANDWIDTH control is operated, the skirts of the equalization curve move in and out but the peak gain remains constant (see Fig. 1). As the TUNING control is operated, the curves in Fig. 1 slide along the frequency axis but their shape is unchanged. If shelving characteristics are desired, they may be approximated by adjusting the lowest band to 20Hz and the highest band to 20kHz. The breakpoint of the shelving characteristic is then adjusted with the BANDWIDTH control.

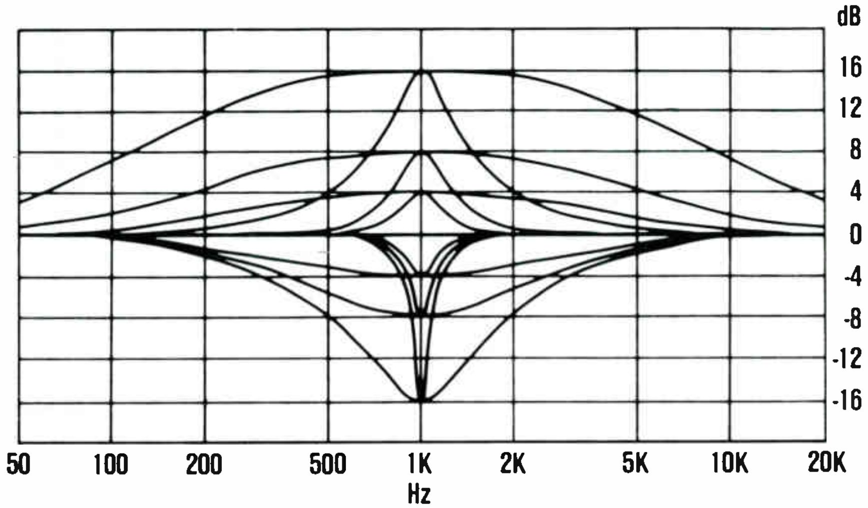


Figure 1

Performance Specifications

Specifications apply to each channel except as noted. All specifications apply when equalizer drives 600 ohm or higher impedances. All noise specifications assume a 20-20,000 Hz bandpass filter with 18 dB/octave Butterworth skirts.

Operating Controls: EQUALIZATION, EQUALIZATION IN/OUT, BANDWIDTH, and TUNING for each of four bands. MASTER EQUALIZATION IN/OUT, GAIN, POWER ON/OFF.
Frequency Response: (EQ controls set mechanically flat) ±0.25 dB, 20-20,000 Hz.
Available Gain: + 12 dB, adjustable to -∞ by means of front-panel GAIN control.
Input: (RF suppressed)
Impedance: (each leg) 100K in parallel with 1000pF, electronically balanced. Driving impedance should be 600 ohms or less.
Absolute Overload Point: + 26dBm.
Output: (RF suppressed)
Level: greater than + 19 dBm into 600 ohms, 20-20,000 Hz
Impedance: 47 ohms in parallel with 1000pF, unbalanced. (Option 01 provides a transformer-balanced output for both channels)
Equalizer is unconditionally stable and will not ring with any captive load.
Risetime: less than 4 microseconds.
Slew Rate: greater than 6 V/microsecond. Internal bandlimiting assures that slew rate limiting will not occur with even the most severe equalization and program material.
Square Wave Response: Square wave exhibits no spurious ringing at any output level. The only ringing observable is that theoretically associated with any given equalization curve.
Circuitry: active RC, utilizing FET-input IC opamps. The output line driver utilizes a discrete transistor current booster.
Total Harmonic Distortion (+ 18 dBm output): less than 0.025%, 20-20,000 Hz. Typically less than 0.002% at 1kHz, + 18 dBm.
SMPTE Intermodulation Distortion: Typically 0.008% at + 18 dBm equivalent peak output, using 60 Hz/7 kHz; 4:1.
Noise: At Output, GAIN control adjusted for unity gain, all EQ switches IN, all EQ controls FLAT: Less than -84 dBm; -87 dBm typical.
Overload-to-noise Ratio of Single Parametric Bandpass Filter: greater than 102 dB for any combination of TUNING and BANDWIDTH settings.
Interchannel Crosstalk, 622B dual-channel equalizer: less than -90 dB, 20-20,000 Hz.
Equalization Characteristics: Figure 1 shows curves corresponding to the maximum and minimum bandwidths for each band. DB equalization contributions of the individual bands add without interaction. BANDWIDTH, TUNING, and EQUALIZATION controls are all continuously variable.
Range of Adjustment of "Q": 0.29 to 3.2.
Range of Adjustment of Peak Equalization: + 16 dB to -∞ . Typical notch depth obtainable is 40 dB.
Tuning Range (per band): 20-500 Hz, 68-1700 Hz, 240-5850 Hz, 800-20,000 Hz. Tuning dials are calibrated at ISO preferred frequencies.
Power Requirements: 115/230 volt 50 60 Hz AC, approximately 4 watts (622A), 7 watts (622B). Captive "U-Ground" power cord. Option 02 eliminates the AC power supply. Power requirements for the Option 02 version are ± 18 to 28 volts DC at 60 ma per equalizer channel. Option 02 is supplied on special order only, and is recommended only for users planning to install a large number of 622 channels in a given installation.
Overload Lamp: will light for approximately 200 mS if the instantaneous peak output of any amplifier in the equalizer is driven within 1 dB of its clipping point.
Size: 19" (48.3 cm) wide x 3.5" (8.9 cm) high x 5.2" (13.3 cm) deep.
Shipping Weight: 10 lbs. (4.5 kg).

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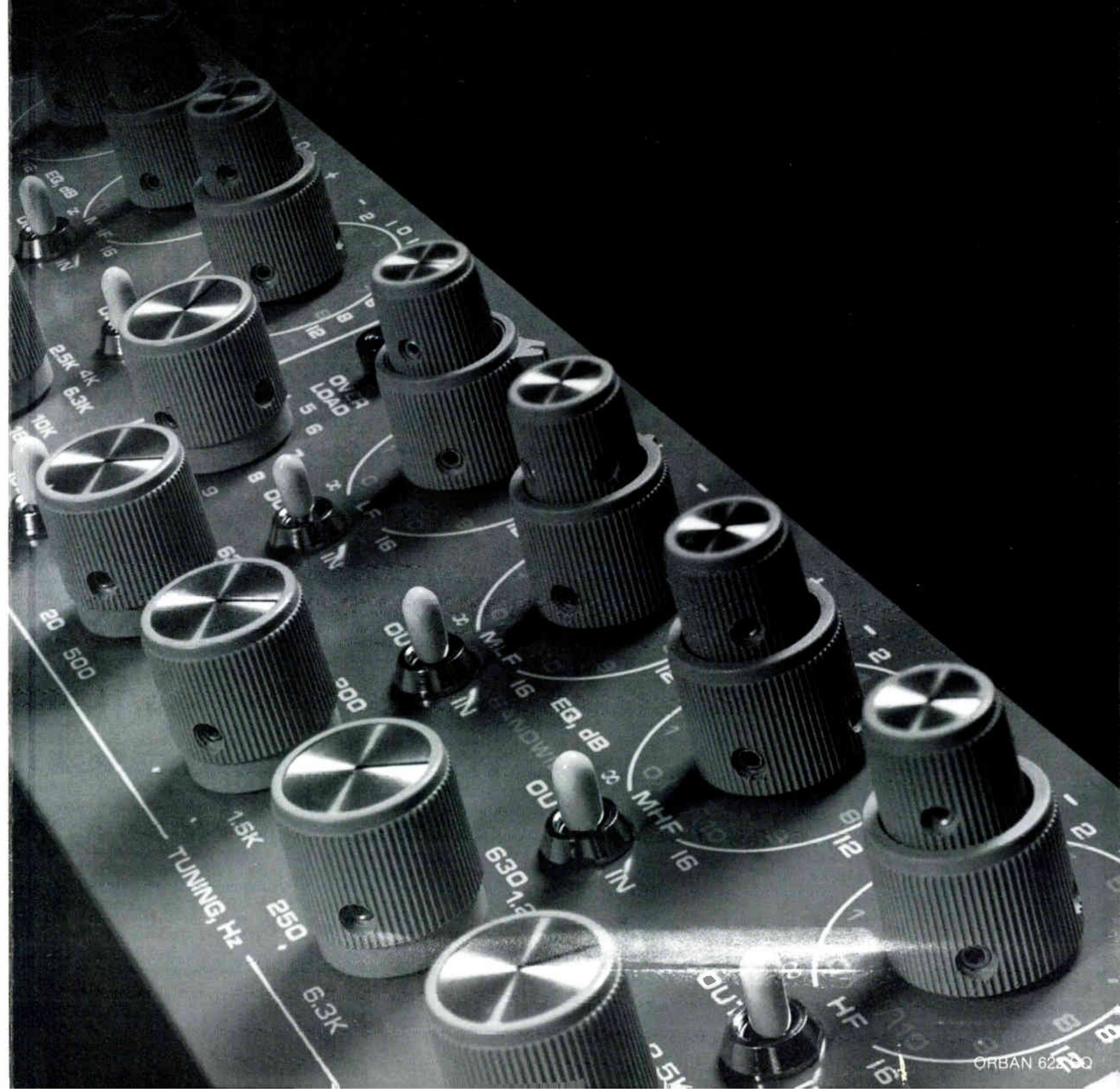
The Orban 622 Parametric Equalizer

The World Class Parametric EQ



Performance Highlights

- +16dB, –infinity dB equalization range
- Each section tunes over 25:1 frequency range
- "Q" adjustable from 0.29 to 3.2
- "Constant-Q" operation enables use of equalizer as notch filter
- True Parametric operation: all controls non-interacting
- Four totally non-interacting peak boost/cut sections, each with TUNING and BANDWIDTH control
- Front panel GAIN control; 12dB gain available
- In/out switches for each section, as well as entire equalizer
- "Peak stretching" overload indicator warns of overload anywhere in equalizer
- Active balanced input; unbalanced output. Transformer-balanced output available
- Very low noise and distortion
- High slew rate for minimum TIM (SID)
- High stability active RC circuitry
- Single or dual-channel models
- RFI suppression of input, output, and power leads
- 115/230V 50-60 Hz AC power supply standard

The 622 Parametric Equalizer

Description
The Orban 622 is a true Parametric Equalizer of high professional quality, providing outstanding versatility and control. The four sections in each channel each use "constant-Q" circuitry. This results in an equalizer of outstanding musicality, and permits any section to be used as a narrowband notch filter (with typically better than 40dB rejection) to effect room tuning or to eliminate fixed-pitch interference, like hum. The sections are totally non-interacting: the total equalization (in dB) is simply the sum of the equalizations of the individual sections.

Considerable attention has been devoted to human engineering, maintainability, and performance in the harsh environments often encountered by the professional user. Levels, impedances, and connectors are fully compatible with virtually all professional equipment. The rugged chassis provides shielding against electrical interference, RFI, and dust. Reliability is assured by a formal burn-in program and additional high temperature burn-in of most semiconductors.

Each feature of the 622 Parametric Equalizer has been thoughtfully chosen and implemented to make the equalizer a particularly powerful tool in nearly all areas of audio: sound reinforcement, public address, recording studio, broadcasting, motion picture sound, disco, theater . . .

Parametric Equalizers—An Explanation
In general, "Parametric" means anything an equalizer manufacturer chooses it to mean. The most commonly accepted definition is that a "Parametric" equalizer provides continuously variable control over the three fundamental *parameters* of equalization: the *amount* of peak or dip (in dB), the *frequency* (the "center frequency") at which the maximum peak (or dip) occurs, and the *bandwidth* (the number of frequencies on either side of the center frequency which are affected by the equalization.)

Bandwidth is a poorly defined parameter for equalizers. (In particular, it is *not*, as often stated, the ratio of the center frequency to the frequency at which the equalization is 3dB down — what if we're using only 2dB eq?!) The bandwidth is related to a more precisely defined factor called the "pole Q", or simply the "Q", for short. If this factor is kept constant as the EQ control is operated, then a curve family called "constant-Q" (see Fig. 1) is created. These curves are not reciprocal —the dip curves are narrower than the boost curves. Experience has shown that these curves produce a more musically-useful equalization with minimum readjustment of the BANDWIDTH control as the EQ control is operated. Moreover, unlike the more common reciprocal curves, they permit the creation of deep narrowband notches which are highly useful for suppressing sounds of fixed pitch, with negligible degradation of the rest of the program.

There are two fundamentally different types of Parametric Equalizer. Orban manufactures both types. The 622 is a "true Parametric." This means that adjustment of a single parameter (like the center frequency) does not affect the other two parameters. This configuration is preferred when maximum convenience is desired. Conversely, a "quasi-Parametric" (like our 672A) permits some interaction (usually changing center frequency also changes "Q") to achieve lower cost. For more detailed information on these important but challenging subjects, please request our free paper "How To Choose Equalizers For Professional Applications" by Robert Orban.

Regardless of configuration, Parametric Equalizers are usually superior to other types when maximum control, flexibility, and freedom from undesired side-effects are desired. In using non-parametric equalizers, you must live with whatever bandwidth and whatever discrete center frequencies the manufacturer has chosen. And you don't have any control over how the bandwidth varies as you change EQ. In graphic equalizers, large boosts over a broad bandwidth often become excessively colored and ringy compared with the results obtainable from an optimally adjusted Parametric.

Applications

Sound Reinforcement
The 622 can often do a surprisingly effective job of "tuning" a sound reinforcement system to a room. The availability of four narrowband notches means that sharp resonances can be dealt with—often more effectively than with third-octave filter sets with fixed filter frequencies. While not designed to replace third-octave filters, the 622 can often augment their effectiveness and in many cases can make surprisingly substantial improvements all by itself. One useful variation is to use both equalizers in the 622B in series—one to notch out feedback and one to provide broadband equalization.

In large scale reinforcement systems for traveling shows, the Orban Parametric is highly useful in equalizing stage monitor systems. In bi-amped and tri-amped installations, the use of one channel of Parametric equalization after each output of the electronic crossover has proven to be of substantial value in optimizing the performance of the individual drivers in the loudspeaker system. Anywhere a conventional equalizer is used, the 622 can do the job better. If more complicated equalization is required, use several channels in cascade. The noise level is low enough to permit this.

Motion Picture Sound
The 622 is an ideal replacement for the graphic equalizer ordinarily used for dialogue equalization. The mixer gets finer control, plus the ability to instantly notch out the extraneous sounds that always seem to plague location recordings. In the music recording studio, the 622's improved adjustability means better sound in the theater. In production, use it for special effects like telephone or "old time" recordings.

Recording Studios
Every recording studio needs at least a few channels of Parametric equalization to handle the tough chores that the internal console equalizers can't deal with. Many experienced Orban Parametric users prefer to have one channel of Parametric equalization on each console input. With practice they're fast and easy to use, and the powerful features (like notching and fine-tuning are instantly available without patching).

The 622 is a particularly valuable adjunct to an electronic music synthesizer—you can create high "Q" formants and shape the spectrum so that the sound comes alive.

If you need to correct the equalization of a finished track because of second thoughts after the mix, the 622 can create the finishing touches as no ordinary equalizer can. It's better than even a third-octave graphic, because the 622 can create broad, non-ringing boosts—the graphic is much more colored and ringy.

Broadcasting
Use the 622 in the production studio to enhance the announce mike and to create special production effects that make your station stand out among its competitors. Meanwhile, another 622 can be quietly and efficiently equalizing the program line for maximum punch and brightness on the air.

Use the 622 to equalize phone and remote lines for flat response—it's much more versatile than the standard phone company equalizers. In the main studio, use it on the announce mike to equalize for maximum presence and also to notch out undesired sounds like mechanical hum from cart machine motors or air conditioning noise. Whatever your application, the 622's RFI suppression and optional balanced-output transformer mean trouble-free installation even in high-RF environments.

In Summary

Many people are now aware of the *power* of parametric equalization: the almost sensual satisfaction of getting the sound exactly *right*. These same people are also demanding professionals, insisting on inaudible noise and distortion, human engineering, quality "feel," and uncompromising reliability. We at Orban feel that there is no cheaper equalizer that delivers this full degree of professionalism, and no more expensive equalizer which provides an improvement in performance proportionate to its cost. Our 622 is also backed up by a company which is firmly established in the industry and is committed to service, stability, and responsiveness to customer needs. That's why our 622 is such a fine choice for any professional who needs an equalizer.

The 622 is available from your local Orban professional audio dealer. Call or write for the name of the dealer nearest you.

orban

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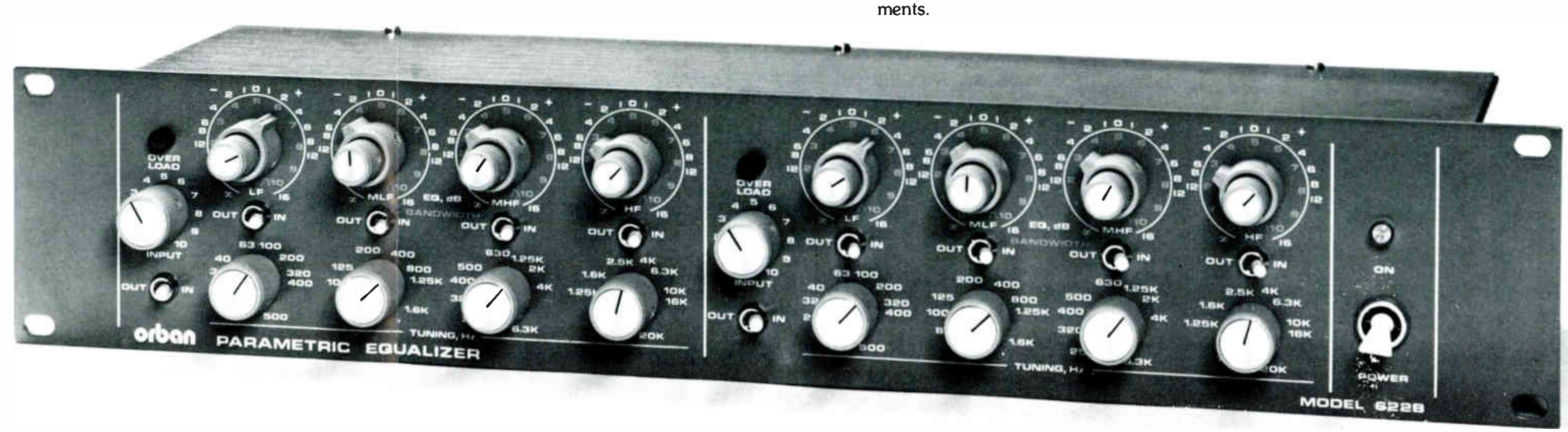

Broadcasting

The aggressive compression and limiting employed by many broadcasters in an effort to win audience share may result in a substantial exaggeration of announcer sibilance. In addition, there are some "problem announcers" with unnaturally sibilant voices whose sibilance may be unacceptably aggravated by even moderate audio processing and/or the choice of microphones.

Such problems can be solved with the 536A. It can be used beneficially by itself or in a complete signal processing chain for voice only, aided by equalization (like the Orban 622 or 672A parametric equalizers). The extensive RFI protection means easy installation in the broadcast environment.

The 536A can also be used in TV newsrooms and production facilities to combat the sibilance problems often encountered when using "tie-tac" mics. Many female announcers' voices are highly sibilant; the 536A can add a more naturally listenable quality to these voices.

Summary

Virtually everyone involved in professional sound needs de-essing—often at a moment's notice! Either you must deal with at least one "problem voice" which requires de-essing to reach contemporary standards of recorded quality, or sibilance buildup problems will hold you back in your quest to transcend the merely adequate and achieve the truly excellent.

If you recognize either problem, the Orban 536A is your ideal solution. It's clean. . . it's quiet. . . and it's **simple.** It's a **problem solver,** not a problem causer. See your dealer about the new 536A two-channel, rack-mounted de-esser from Orban—the de-essing expert.

orban

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Specifications

Frequency Response: ±0.25dB, 20-20,000Hz

Total Harmonic Distortion: (de-essing defeated): <0.025%, 20-20,000Hz, @ +24dBm

Total Harmonic Distortion: (de-essing in): <0.5% @6kHz

Output Noise: (20kHz bandwidth): < −75dBm. Dynamic range from noise floor to clipping exceeds 100dB.

Input Level Variation for Constant De-essing: >15dB

Input Characteristics:
Impedance: >10,000 ohms, active-balanced bridging
Level: −10 or +4dBm (strappable)

Gain: +20dB or +6dB (Dependent on input strap; referred to balanced output. Referred to unbalanced output, gains are 6dB lower.)

Output Characteristics:
Impedance: Approximately 100 ohms, active-balanced-to-ground. Fully-floating transformer output optional. Unbalanced output available from either output to ground.
Level: Drive capability into 600 ohms exceeds +25dBm, 20-20,000 Hz

Crosstalk: < −80dB

Attack Time: approximately 1 ms

Recovery Time: approximately 10 ms

Variable-Gain Element: junction field-effect transistor

Indicators:
Two-element LED gain reduction meter;
LED OVERLOAD indicator
LED POWER ON indicator

Operating Controls:
THRESHOLD control (each channel)
DE-ESSING IN/OUT switch (each channel)
POWER ON/OFF switch

Connectors: All inputs and outputs appear on Jones 140-Y-type barrier strip (#5 screw). Chassis is punched for optional installation of paralleling XLR-type input and output connectors.

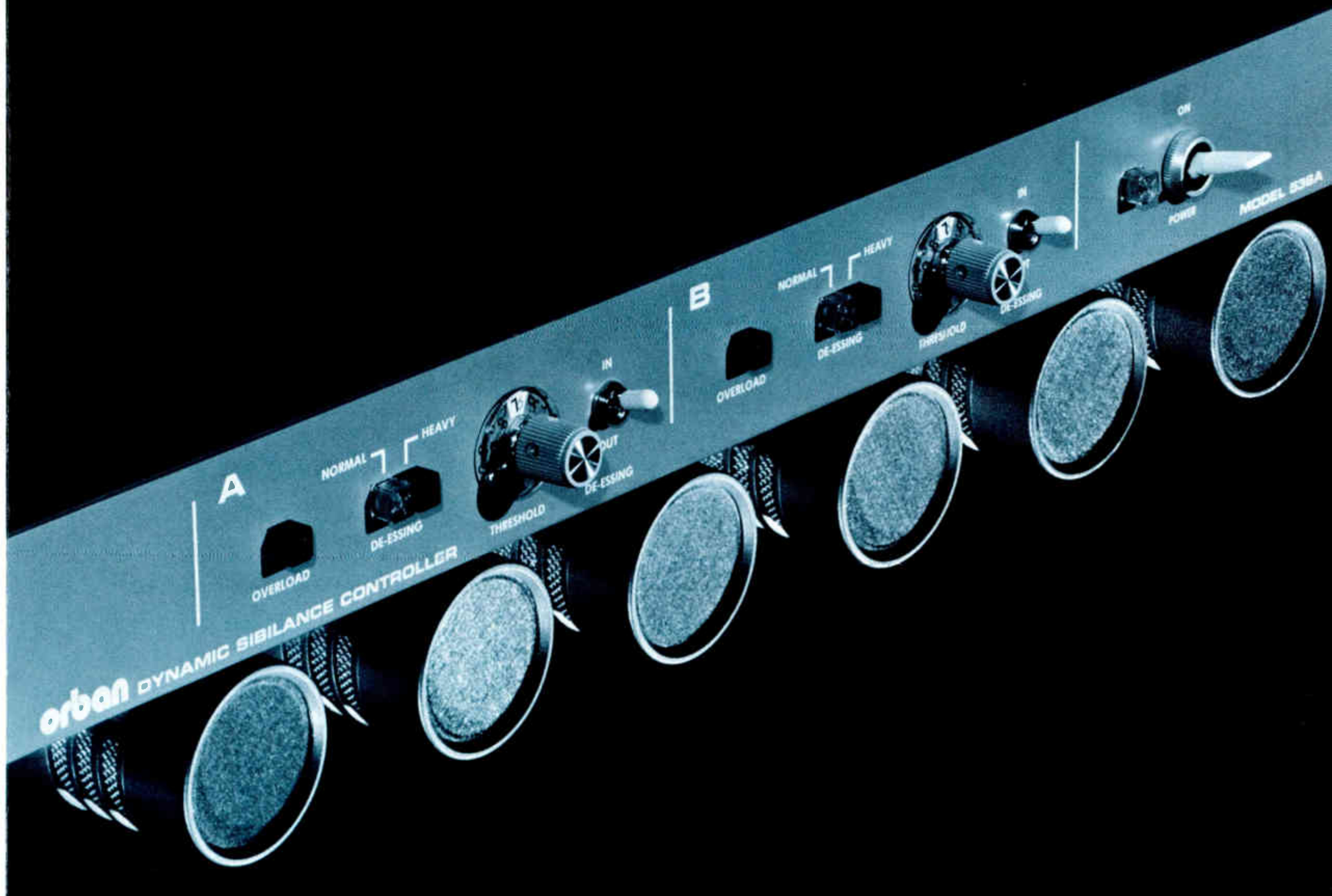
Power Requirements: 115-230 volts AC, ±10%, 50-60Hz, approx. 6 watts. Captive power cord with "U-Ground" plug to United States standards.

Size: 19" (48.3cm) wide x $1\frac{3}{4}$" (4.45cm) high x $5\frac{3}{4}$" (14.6cm) deep

Weight:
Shipping: 7 lbs. (5.2 kg)
Net (without options): 5 lbs. (2.3 kg)

The Orban 536A Dynamic Sibilance Controller

Two channels of de-essing at an affordable price.



Performance Highlights

- ☐Two independent channels
- ☐Effective, inaudible de-essing over a 15dB input range
- ☐Active-balanced input, strappable for +4 or −10dBm
- ☐Active-balanced output with transformer option
- ☐Dual-LED gain reduction metering
- ☐Overload/noise ratio typically 105dB
- ☐Very low distortion
- ☐Effective RF suppression
- ☐Top-quality professional construction
- ☐Cost-effective 19" rack mount package
- ☐Low cost-per-channel

Related Products:
526A Single-Channel Dynamic Sibilance Controller (with switchable mic/line inputs)

ORDER GUIDE

536A Two-Channel Dynamic Sibilance Controller
Options:
RET-22 XLR-type connectors, kit
RET-23 Transformer-coupled outputs(500/600 ohms), kit
Note: RET- kits are factory-installed if requested, or may be ordered later as field-retrofit kits

Description

The new Orban 536A Dynamic Sibilance Controller is an improved, two-channel version of our popular single-channel 526A (but without the mic input). The 526A and its ancestor, the 516EC, are considered to be the industry-standard de-essers and are found in professional audio facilities throughout the world. Now, with the introduction of the 536A, we offer the most cost-effective package available today—two channels of de-essing at an affordable price.

Like its predecessors, the 536A is designed to work effectively on **voice only** in its many professional applications. (Users requiring de-essing of mixed tracks should consider the Orban 422A/424A Gated Compressor/Limiter/De-esser.)

Compared to its competitors, the 536A offers vastly simpler setup, improved noise and distortion performance, and no emphasis of residual IM distortion while de-essing is occurring.

The 536A is active-balanced at both inputs and outputs. For applications requiring a fully-floating output, optional output transformers can be fitted. Like all Orban products, the 536A features RF filtering of input, output, and power leads to help assure trouble-free installation in high RF fields.

The 536A is simplicity itself to operate. There is only one adjustment: a THRESHOLD control. The amount of sibilance energy is limited to a certain **fraction** of the non-sibilant speech energy—even if the input level changes as much as 15dB. The THRESHOLD control determines this fraction. Thus, a consistent balance between the "voiced" and "sibilant" sections of speech is achieved whether or not input levels are well-controlled. This is especially useful with off-mic voices, or in any application (such as motion picture sound) where speech levels vary widely.

A dual-LED display indicates the amount of de-essing which is occurring ("Normal" and "Heavy"), warning of possible misadjustment of the THRESHOLD control. An peak-stretching OVERLOAD indicator warns of clipping anywhere in the circuit.

The IN/OUT switch can be operated at any time without clicks, pops, or gain changes.

Critical parameters, such as attack time, release time, and the frequency characteristics of the control loop, have been preset after extensive listening tests on many voices. Therefore, these parameters can never be misadjusted by the user. Instead, the 536A is simply **there**—ready to do the job quickly and efficiently with minimal hassle.

How It Works

"Esses" are detected by means of a sharply selective filter which effectively discriminates between "ess" frequencies (centered around 6kHz) and lower-frequency vocal components. In addition, the absolute threshold at which de-essing begins to occur is automatically forced to track the peak input level. This threshold is thus a constant **fraction** of the input level, resulting in **constant de-essing action over a wide range of input levels.** (The fraction is determined by the setting of the THRESHOLD control).

When an "ess" attempts to exceed the automatically-varying threshold, the 536A decreases its gain as necessary to reduce the "ess" level back to the threshold. Recovery is so quick that the following vocal sound is not audibly affected. Because the gain of the **entire** channel is reduced (not just the high frequencies), any IM distortion existing on the original recording medium is reduced along with the "ess", and coloration of the "ess" sound does not occur.

Non-linear control voltage smoothing assures that excessive modulation distortion will not occur under de-essing conditions despite the extremely fast (approximately 10ms) release time employed.

The variable-gain device is a junction FET, assuring a gentle overload characteristic and freedom from control-voltage leakage into the signal.

Limitations

There are some limitations: First, using the 536A on voice which has already been mixed with other sounds is often unsuccessful because it can mistake sounds having high frequency content for sibilance, with unpredictable results. If de-essing of mixed vocals and instruments is desired, we recommend that you evaluate our Model 422A (mono) or 424A (stereo) Gated Compressor/Limiter/De-essers which can perform that task.

Another limitation is that, because the 536A uses frequencies of 6kHz and above to control its action, the minimum bandwidth required of the source material is approximately 8kHz. Occasionally, bandwidth problems arise when the "sibilance" is mostly IM distortion caused by tape overload (particularly with cassettes), by use of telephone-quality carbon microphones, or by attempts to de-ess recordings distorted by previous transfer to optical film. If the sibilance exists as IM without significant energy above 6kHz, then the de-esser cannot detect and control it.

The track must also be reasonably free of noise or hiss in the region above 6kHz. Normally these factors do not cause problems except in the case of the lowest-quality program material.

It is known that some dialects of non-English languages can trigger de-essing action on sounds other than sibilance. This is ordinarily not a problem if the THRESHOLD control is set carefully.

Application Ideas

Recording Studio

Current vocal equalization practices in the pop recording industry tend to include large amounts of high frequency boost to increase presence and articulation. These boosts, particularly when combined with limiting or compression, can boost sibilance to unpleasantly high levels. Using the 536A **after** the equalizer and limiter can reduce the sibilance to levels that sound natural and right. With the 536A, equalization and limiting are no longer constrained by the problems of unnatural sibilance levels, and may therefore safely be used to achieve the ultimate artistic goals of the producer and engineer. Use of 536A during recording allows the producer/engineer to get the intimate, "tight to the mic" sound that has become so popular on hit records while substantially reducing the probability of disk overcutting or cassette saturation when the master is transferred to these common consumer media.

The availability of two independent channels of de-essing in the same package means that two different voices can be de-essed simultaneously without interaction, making the 536A extremely cost-effective.

Motion Picture Sound/Video Production

The susceptibility of the variable-area optical soundtrack to high frequency crossmodulation distortion made de-essing compulsory in motion picture recording. The 536A can fulfill this need in a simple, natural, and inaudible way—with solid-state reliability and with the fully balanced input and output interface requirements characteristic of motion picture re-recording theaters.

Similarly, in video post-productions, there are often problems in getting properly-balanced dialogue tracks—particularly when mic placement was compromised by the needs of the video! The 536A allows radical corrective re-equalization of such problem tracks without overload due to sibilance.

Sound Reinforcement

Ultra-close miking is the norm in most amplified music. If substantial 6kHz boost has been applied to increase presence—either by means of an equalizer, or by use of a mic with a built-in "presence peak"—then sibilance and "spitting" can become a real problem. The 536A is a cost-effective solution. Up to two microphone channels can be de-essed simultaneously with minimal setup problems.

PRELIMINARY (Subject to Change)

orban

The New Orban 222A Stereo Spatial Enhancer

An effective and affordable enhancement tool for on-air spatial definition.

- Proprietary, patent-pending technique detects and enhances psychoacoustic directional cues which are present in all stereo program material.
- Increases brightness, impact, and definition of music.
- Front-panel ENHANCEMENT and WIDTH LIMIT controls allow tailoring of processing to user requirements.
- No increase in FM multipath distortion, no unnatural exaggeration of reverberation, and no increase in sensitivity to vertical tracing distortion in disc playback.
- Full mono compatibility.
- Complements any broadcast audio processor without changing the station's "sound".
- Easy-to-read LED bargraph displays indicate status and degree of enhancement.

Why Stereo Spatial Enhancement?

In contemporary broadcast audio processing, high value is placed on the loudness and impact of a station compared to its competition. Our industry-leading OPTIMOD™ processors already have made a major contribution to competitiveness.

Now, the new Orban 222A Stereo Spatial Enhancer can augment your station's **spatial** image the way your audio processor maximizes loudness and brightness. Your stereo image will become magnified and intensified; your listeners will also hear more loudness, brightness, dynamics, and depth.

How You Use It & What It Does

The 222A has stereo inputs and outputs. It is designed to be inserted in the program line at the studio prior to processing. It dynamically changes the amplitude and phase content of the program material to increase apparent width, depth, and transient definition. It does not add "musical distortion". Intelligent gating makes the unit immune to small errors in channel balance, prevents over-enhancement, and avoids the "mushy", homogenized sound that has so often been the result with earlier techniques.

A Competitive Edge Without Nasty Side-Effects

There have been many attempts to process stereo signals to increase width and dimensionality by altering the ratio of the L–R signal to the L+R, or by adding controlled distortion (which sounds good on some program material but not on all.) The major problem has been that any enhancement of the L–R signal by prior techniques has always resulted in a significant increase in multipath distortion.

The 222A does not increase multipath distortion, exaggerate reverberation, or increase sensitivity to vertical tracing distortion in disc playback. It can be placed in the program line confidently with the assurance that it will provide the benefits of stereo spatial enhancement without the by-products which cause listener annoyance and tune-outs.

The Orban 222A Stereo Spatial Enhancer uses a new patent-pending technique which has been **extensively tested** in real-life, high multipath environments. These tests have resulted in uniformly rave reviews.

Most of All — Affordable Enhancement

In the refinement of your station's audio processing, you've no doubt become wary of equipment that's over-hyped, over-priced, and under-powered. The 222A Stereo Spatial Enchancer is **affordable**: It's price is only $995.00*. And it works.

Find out how well it works by trying it on the air, with your own format and your own processing chain. Contact your preferred Orban Dealer and arrange for a demo when the 222A becomes available. We think you'll find that our new enhancement tool is a powerful, competitive weapon that's just right for your image.

Estimated Price: $995.00*
(* Suggested List Price, USA)

First Delivery: January, 1988

Orban Associates Inc. 645 Bryant Street San Francisco, CA 94107
(415) 957-1067 Telex: 17-1480 FAX: (415) 957-1070

PRELIMINARY (Subject to Change)

orban

The New Orban 642B Parametric Equalizer

The Next Generation of Parametric Excellence

- 4-band, dual-channel parametric equalizer. Each section offers overlapping tuning with a 25:1 frequency range; +16dB boost/−40dB cut in each band.
- Bandwidth variable from about 5 to 1/4 octaves: ("Q": 0.29 to 5.0).
- "Constant-Q" design enables use of equalizer as a true infinite-depth notch filter; vernier frequency control on each band facilitates precise tuning of notches.
- Tunable 18dB/octave high-pass filter and 12dB/octave "Automatic Sliding Besselworth"™ low-pass filter provide maximum flexibility while preserving musicality.
- Front-panel **Cascade** switch permits use as either a two-channel 4-band, or one-channel 8-band equalizer.
- In/out switches on each band and on each channel simplify comparison of EQ settings with flat.
- 12dB make-up gain is available.
- Overload indicator warns of overload anywhere in equalizer.
- Noise and distortion specs significantly better than 16-bit digital.
- Active balanced inputs and outputs with optional output transformer.
- Main signal path is free of coupling capacitors.

Proven Orban EQ Technology Takes a Giant Step

In 1974, we introduced our 621-series parametric equalizer. It became a classic. It was followed by the very popular 622 series. Now, the demands of the digital era and improvements in components have inspired us to bring forth a new problem-solver, the 642B Parametric Equalizer.

Orban has espoused the virtues of parametric equalizers over graphics for nearly fifteen years. The audio world has gradually changed. Parametrics have become the first choice for solving a wide range of EQ problems where graphic equalizers were previously favored.

In an effort to provide a more perfect equalizer, Orban has built on the many strengths of the 622B, improving circuitry and adding important new features. **In terms of musicality and control, the 642B behaves like the widely-used 622 series — and its measured performance is significantly better.**

Full-Featured Parametric Power

The Orban 642B Parametric Equalizer is an efficient and powerful unit that allows for a wide scope of both creative and corrective enhancement. Our "constant-Q" design avoids interaction between operating parameters so that the alteration of one parameter (such as frequency) does not subsequently change another (such as bandwidth). One specific benefit of "constant-Q" design is that it enables the use of the 642B as a true notch filter/feedback suppressor with infinite-depth, narrowband notching capability.

In order to make the notch filtering capability of the 642B even more efficient, Orban has introduced a vernier control which varies the center frequency ±10%, thus allowing the user to quickly tune the equalizer to an exact frequency null.

Another important new feature of the 642B is a front-panel **Cascade** switch. This allows quick conversion of the unit to be used as either a two-channel 4-band or single-channel 8-band equalizer.

For further flexibility, a continuously-variable 18dB/octave, third order, Butterworth high-pass filter and 12dB/octave "Automatic Sliding Besselworth" low-pass filter are incorporated. The "Automatic Sliding Besselworth" filter is a proprietary Orban design that gracefully changes from a smooth, musical-sounding, Bessel characteristic to a sharper Butterworth one as the cutoff frequency is increased. This results in a filter that always removes the maximum amount of unwanted noise while simultaneously avoiding harshness.

Improvements in technology have allowed us to enhance the noise and distortion specifications of the 642B so that they are significantly better than 16-bit digital. The 642B boasts an overload/noise ratio of better than 110dB, making it the equalizer for contemporary, low-noise recording. Modern servo-stabilized design has allowed the elimination of all capacitors from the signal path. Fully balanced inputs and outputs, extensive RF suppression, and well-known Orban reliability are also featured.

Specific Applications

Recording Studios: As much as console equalizers have improved over the years, every quality studio requires full-featured, high-function outboard equalizers to deal with unusual problems that can't be properly resolved by on-board EQ. The 642B offers superior musicality as a broadband, shaping equalizer and functions gracefully as a vernier-tuned notch filter. The 642B can easily be patched into any track to quickly solve bothersome problems. In addition, tunable HP and LP filters add flexibility for special effects or for noise reduction.

Sound Reinforcement: The fully parametric 642B can do a superb job of tuning a sound reinforcement system, usually with far better results and in less time than with a graphic equalizer. The theoretical and practical limitations of graphics are well known. Parametrics are a better choice for tuning quality systems since their filter design minimizes phase shift, noise build-up, and ringing. In addition, they provide greater flexibility to deal with room resonance problems without creating troublesome artifacts.

High and low-pass filters allow shaping the passband so that power is not wasted feeding the drivers signals they can't reproduce.

Infinite-depth, tunable notching makes the 642B the perfect choice for monitors as well. The addition of a vernier control allows the user to more quickly find the exact frequency that is "howling" and to notch it instantly.

In short, anywhere that a conventional graphic equalizer is being used, the 642B can probably do the job better. The flexibility and musicality of the Orban parametric remains unsurpassed by any other equalizer.

Video and Film Post-Production: Mixing dialogue, music, and effects can be a brutally demanding, time-pressured process. And it's typically the time when recording errors are "discovered".

At least one 642B belongs in a rack nearby to quickly patch into problem tracks. Its notching and bandpass filtering capabilities allow for foolproof suppression of unwanted sounds such as air-conditioner rumble, camera whine, and other intrusions. The 642B can simultaneously be used to fatten-up or enrich dialog and effects for maximum punch. Flexible HP and LP filters can effectively remove broadband noise at the frequency extremes.

Broadcasting: The 642B will be a welcome addition to your production studio to enliven in-house promos, spots, and ID's. Meanwhile, another 642B can quietly equalize the program line for maximum punch and brightness, or EQ phone lines for flat response. In the main studio, the 642B can equalize the announce mic for maximum presence and punch, and remove mechanical hum and noise from cart machines or air conditioners. Whatever your application, the 642B's rigorous RF suppression means trouble-free installation, even in high RF environments.

The Next Generation of Parametric Excellence

We have avoided the temptation to overbuild the 642B: You don't have to pay for unnecessary bands, for esoteric, expensive components that don't provide audible improvements in performance, or for features that sound enticing but are difficult to use. The Orban 642B Parametric Equalizer is an elegant new tool that is sure to become the "ultimate equalizer" for all audio professionals.

Contact your preferred Orban Dealer for more information about the new 642B.

Projected Price: $1,200

Projected First Delivery: Early Spring '88

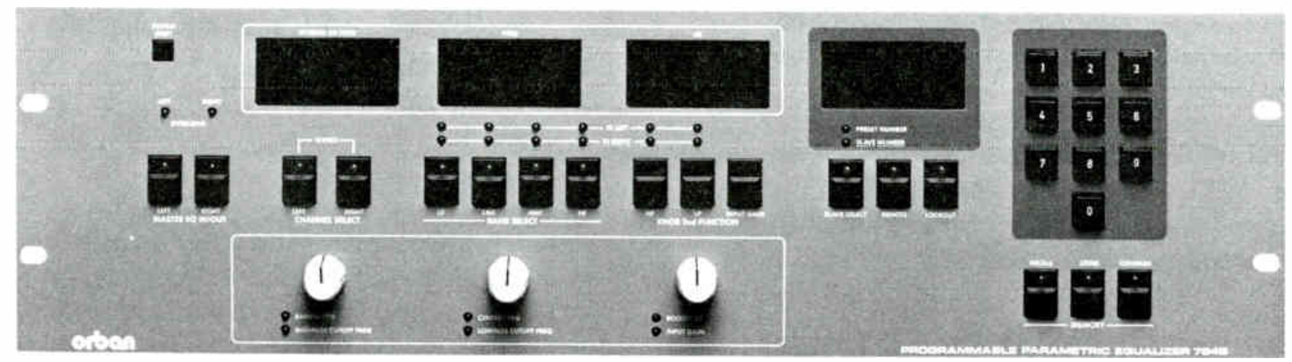
Orban Associates Inc. 645 Bryant Street San Francisco, CA 94107
(415) 957-1067 Telex: 17-1480 FAX: (415) 957-1070

800055-000-01 10/87

PRELIMINARY (Subject to Change)

The New Orban 764B Programmable Parametric Equalizer

A major advancement in EQ technology



The Orban 764B Programmable Parametric Equalizer represents the most powerful equalization tool available for audio professionals — one that will vastly simplify many laborious recording and mixing functions and permit accurate recreation of specific EQ control set-ups.

- 4-band, dual-channel parametric equalizer. Each section offers overlapping tuning having a 25:1 frequency range; +16dB boost/−40dB cut in each band.
- Bandwidth variable from 5 to 0.1 octaves ("Q": 0.3 to 15).
- Proven Orban "Constant-Q" design enables use of equalizer as a true notch filter.
- Tunable 18dB/octave high-pass filter and 12dB/octave "Automatic Sliding Besselworth"™ low-pass filter provide maximum flexibility while preserving musicality.
- 99 non-volatile memory registers for instantaneous storage and accurate recall of complete control setups, including input gain.
- Digital displays show current settings of control parameters.
- High-quality, no-compromise audio path having no VCA's.
- Up to 14 two-channel slave units can be addressed by the master unit. Each pair of channels can be ganged to track in stereo or can be programmed independently.
- MIDI-controllable. Port for remote control. Provision for future serial interfaces.

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800054-000-01 10/87

The Evolution of Programmable EQ

Manual equalizers (such as our 642B Parametric Equalizer) are perfect for many corrective and creative equalization chores, yet there is often a need to make instantaneous EQ changes during a session or live event. The ability to quickly and accurately recall EQ curves that have proven successful in prior work or in other venues is also quite desirable.

The Orban 764B Programmable Parametric Equalizer represents a major breakthrough in contemporary equalization technology, offering unsurpassed flexibility with full storage and recall capability. All parameters — frequency, bandwidth, boost/cut, high and lowpass filters, and input gain — can be modified to specific user requirements and stored and recalled at will. Flexible remote control with provisions for MIDI and future serial interfaces further extend the capabilities of this state-of-the-art programmable processor.

Fully Programmable EQ Power

Programming the 764B is simple: just select one of the four bands, then use the rotary controls to alter each parameter. Digital and LED displays show the value of each parameter being adjusted. All four EQ bands, high and lowpass filter cutoff frequencies, input gain, and dual-mono or stereo status can be programmed in any one of 99 non-volatile memory registers. One memory register can be used to store a complete two-channel setup by using a simple three keystroke sequence on the keypad.

Recall of stored set-ups is just as simple and can be effected manually, through MIDI, or via future serial interfaces.

Once setups are stored, the 764B protects your work. You can enter a security lockout code to prevent unauthorized tampering. An internal lithium battery protects memory from power failures for up to five years.

Outstanding Quality Equalization

While programmability is the key feature of the 764B, the heart of the unit is the equalizer itself. Orban's proven "constant-Q" design provides equalization that is smooth and musical. It also allows the unit to be used as a true narrowband notch filter, a broadband "shaping" equalizer, or both simultaneously.

One unique innovation of the 764B is the incorporation of a continuously-variable 18dB/octave, third order, Butterworth highpass filter and a 12dB/octave "Automatic Sliding Besselworth" filter. This proprietary Orban design smoothly changes from a musical-sounding, Bessel characteristic to a sharper Butterworth one as the cutoff frequency is increased. This results in a filter that always removes the maximum amount of unwanted noise while simultaneously avoiding harshness.

Applications

Multi-Track Recording and Mixdown: The Orban Programmable Parametric Equalizer offers the engineer/producer the capability to efficiently obtain repeatable EQ setups day-after-day, even in different studios. Little time need be wasted trying to recreate previously successful effects. Instead, sessions can begin quickly, and energy can be directed towards getting top creative results rather than laboriously tweaking equalizers.

For the studio that desires fully programmable EQ on each channel of a multi-track console, as many as 28 independent slave channels (or 14 stereo pairs) can be stacked in tandem, controlled by the 764B as the master control unit. This array is a powerful addition to any high-quality console and brings it closer to the capability of multi-hundred thousand dollar consoles.

Video and Film Post-Production: Post-production EQ chores can be handled smoothly and efficiently through the use of the 764B. EQ changes for an entire production can be stored in the 99 memory registers and recalled via SMPTE-to-MIDI converter or contact closure pulse. ADR work can be enhanced and made more efficient by storing "personality settings" for commonly-used talent to obtain consistent, repeatable results in

every session. The 764B can also dramatically accelerate Foley work by allowing for the convenient recall of commonly-used EQ settings.

Installed Sound Systems, Touring Sound, or Live Theater: The 764B is an efficient and powerful tool for room equalization (using a real-time, TEF®, or FFT analyzer). The 764B can be used to shape and store a house curve, or series of curves, safely secured with a lockout code. This will assure the contractor or consultant of tamper-proof operation after the job is signed-off.

The full-parametric/notch filtering capability of the 764B offers distinct advantages over traditional 1/3 octave graphics, while avoiding the typical artifacts found in many graphic EQ's (such as excessive phase shift and ringing).

For touring sound companies, the 764B will be a welcome alternative to totally re-equalizing the system for each new venue. The selection of the desired house curve will provide, at the very least, an excellent starting point for any minor readjustments that might be required. If used as a dual-channel unit, one channel can provide the house curve, with the other channel supplying notch filtering.

The 764B is particularly useful for EQing live events such as theatrical shows since the programmability feature allows for quick and easy EQ changes following your automation. Additional slave channels can be easily inserted into individual channels of the mixing console to solve various instrumental and vocal problems.

Disc and CD Mastering: The need for programmable parametric EQ has been apparent for a long time in mastering. The introduction of the high-quality, low-noise and distortion, VCA-less 764B is the ultimate cost-effective solution. The master unit can be used for the program channel, with a slave unit dedicated to the preview channel. EQ requirements for the entire disc can be pre-programmed and easily accessed.

downloading of stored EQ settings can be accomplished through MIDI or serial interface, thus the unit can be routinely made available for use on a future project without losing control set-ups from a previous one.

Flexible Interfaces

The 764B is provided with a remote control port so that a contact closure pulse can call stored setups in sequence from memory. This same port can be utilized in conjunction with a customized remote control system that might be implemented.

MIDI control of the 764B is provided as an option so that the unit can communicate with synthesizers and other MIDI-controlled outboard devices. The external integration of a SMPTE-to-MIDI converter will allow use of the unit with SMPTE time code.

Future integration of a serial interface is also planned as a field-installable option.

Summary: Orban's new 764B Programmable Parametric Equalizer offers unprecedented flexibility to deal with a wide range of problems. It represents a breakthrough in efficiency and performance that will be appreciated in many professional applications.

Specifications:

Frequency:

Band 1: 20-510Hz Band 2: 120-2.53kHz
Band 3: 240-5.06kHz Band 4: 640-20.2kHz

Bandwidth: 0.1 - 5 octaves

Boost/Cut: +16dB boost/−40dB cut in each band

High Pass Filter: 18dB/octave, "Automatic Sliding Besselworth" (20-510Hz)

Low Pass Filter: 12dB/octave, Butterworth (640-20.2kHz)

Gain: +10 to −20dB

Projected Price: Under $2,500
($1,000 per additional 2-channel slave unit)

First Delivery: Spring '88

orban

ORDERING GUIDE & SUGGESTED LIST PRICES

PROFESSIONAL AUDIO PRODUCTS

Revision 17; Effective 15 October 1987
Changes: 464A price increase.
Add 787A and 222A.

Model	Description	Suggested List
111B/1	Spring Reverberation (2 channels)	$899.00
222A	Stereo Spatial Enhancer	$995.00
245F	Stereo Synthesizer	$399.00
275A	Automatic Stereo Synthesizer	$1,895.00
275A/RC	Remote Control for 275A	$295.00
412A	Compressor/Limiter (1 channel)	$425.00
414A	Compressor/Limiter (2 channels)	$799.00
422A	Gated Compressor/Limiter/De-Esser (1 channel)	$629.00
424A	Gated Compressor/Limiter/De-Esser (2 channels)	$989.00
464A	"Co-Operator" - Gated Leveler/Compressor/HF Limiter/ Peak Clipper (2 ch)	$1,195.00
536A	Dynamic Sibilance Controller (2 channels)	$539.00
622A	Parametric Equalizer (1 channel)	$569.00
622B	Parametric Equalizer (2 channels)	$879.00
672A	Mono Graphic Parametric Equalizer	$689.00
674A	Stereo Graphic Parametric Equalizer	$1,299.00
787A	Programmable Mic Processor	$1,995.00
787A/SL	Slave Channel for 787A	$995.00

Prices are domestic U.S. only; F.O.B. San Francisco. Prices based on Buyer's acceptance of Orban Standard Terms & Conditions of Sale are subject to change without notice. All units are supplied for 115V, 50/60 Hz operation unless otherwise specified.

See reverse side for accessories and options.

803001-000-17 10/87

Orban Associates Inc. 645 Bryant Street San Francisco, CA 94107
(415) 957-1067 Telex: 17-1480 FAX: (415) 957-1070

PROFESSIONAL AUDIO PRODUCTS ACCESSORIES

ACRYLIC SECURITY COVERS

All security covers are 19" wide. Add suffix in place of xx to specify color. Screws supplied. Fits most EIA-standard panels. 1 1/4" maximum protrusion.

CL	Clear	
BL	Blue transparent	
WH	Opaque White	

		Suggested List
ACC-11xx	1 3/4" panel (1 rack space)	$43.00
ACC-12xx	3 1/2" panel (2 rack spaces)	$45.00
ACC-13xx	5 1/4" panel (3 rack spaces)	$47.00
ACC-14xx	7" panel (4 rack spaces)	$49.00

ACCESSORIES FOR 787A

RET-045	MIDI Option (order one per 787A)	$95.00
RET-046	Jensen Transformer Mic Preamp (order one for each 787A and 787A/SL channel)	$195.00

ACCESSORIES FOR 674A

ACC-03	Plexiglass security cover for filter section controls	$9.00
RET-007	Balanced output transformers (2) for main outputs.	$32.00
RET-008	Balanced output transformers (4) for both outputs.	$64.00
RET-010	TRS phone jacks for inputs & all outputs.	$13.00
RET-012	XLR connectors for inputs & all outputs.	$30.00

ACCESSORIES FOR 672A

RET-006	Balanced output transformer. Order one per output.	$16.00
RET-021	XLR connectors for input and both outputs.	$18.00

ACCESSORIES FOR 622A/622B

RET-005	Balanced output transformer. Order one per output.	$16.00

ACCESSORIES FOR 536A

RET-022	XLR connectors for both inputs and both outputs.	$24.00
RET-023	Balanced output transformers (2) for both outputs.	$32.00

ACCESSORIES FOR 464A

RET-040	XLR connectors for both inputs & both outputs.	$24.00

ACCESSORIES FOR 422A/424A

RET-014	XLR connectors for input and output. (422A)	$12.00
RET-015	XLR connectors for both inputs and both outputs. (424	$24.00

ACCESSORIES FOR 412A/414A

RET-028A	XLR connectors for input and output. (412A)	$12.00
RET-028B	XLR connectors for both inputs and both outputs. (414	$24.00

ACCESSORIES FOR 245F

RET-019	Balanced output transformers (2) for both outputs.	$32.00

PRELIMINARY (Subject to Change)

orban

The New Orban 787A Programmable Mic Processor

Powerful, programmable processing for maximum impact.

Three-band parametric equalizer, compressor, de-esser, noise gate, and compressor gate integrated in a compact, powerful system. Stores up to 32 different control set-ups in memory for instant recall. Designed for mic processing and voice recording, but versatile enough for many other production uses. Optional MIDI interface.

- Line-level input with optional Jensen transformer mic preamp and 48-volt phantom power.
- 3-band parametric equalizer with variable frequency, bandwidth, and boost/cut for precision control.
- Smooth compressor delivers maximum presence and "punch" while maintaining consistent levels.
- Full-function de-esser helps control excessive sibilance.
- Noise gate attenuates noise by up to 25dB; compressor gate prevents rush-ups during pauses (front-panel control selects noise or compressor gate).
- Effects send-and-return with programmable return level simplifies integration of external reverb or "psychoacoustic exciter".
- STORE, RECALL, and COMPARE buttons provide instantaneous access to 32 user-programmed control setups.
- Digital display shows current settings of control parameters.
- Easy-to-read bargraph displays show output and gain reduction levels.
- Memory protected by internal back-up battery.
- Security code locks programming controls to prevent tampering.
- Built-in control connectors for remote control and MIDI or future serial interfaces.
- Provision for second-channel slave unit for dual-mono or stereo operation.

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A Complete Mic Processing System At Your Fingertips

The Orban Programmable Mic Processor integrates a unique combination of processing functions in a **fully-programmable** package. Its flexibility and efficiency are unmatched. It is simply the most powerful audio processing system available, surpassing the quality of the input stages of multi-hundred-thousand-dollar consoles.

In one compact unit, the Programmable Mic Processor has all the processing tools you need to precisely define the sound of DJs, announcers, narrators, singers, or certain instruments — and, when you've got the sound you want, the Programmable Mic Processor remembers the control settings so you can get exactly the same sound tomorrow, next week, or next year with just the push of a button. The Programmable Mic Processor will remember up to 32 complete setups, eliminating the laborious setting up and "tweaking" of an array of non-integrated processing units before each airshift, recording session, or live event.

Processing Power for Maximum Impact

The 787A has standard line-level inputs. Alternatively, low-impedance microphones can be connected directly to the Programmable Mic Processor's optional **low-noise, low-distortion Jensen transformer mic preamp** (with 48-volt phantom power) through rear-panel XLR connectors. Transformer-coupled to ensure RF immunity, this preamp is vastly superior to most console preamps.

Good processing begins with **consistent control of levels**. DJs, announcers, singers, and musicians sometimes stray away from the mic or alter their dynamics enough to cause levels to vary disturbingly. This makes it difficult to control presence and punch.

The compressor section of the Programmable Mic Processor automatically controls levels. Gain reduction is smooth, unobtrusive, and free from processing artifacts (such as pumping and breathing). The programmable INPUT ATTENUATION control sets gain reduction levels. To facilitate operation, compression ratio and attack time are automatically controlled by the audio signal. For maximum flexibility in shaping voice and instrument characteristics, the Programmable Mic Processor's compression release time can be controlled by the operator.

The Programmable Mic Processor provides **two types of gating** to maximize flexibility and utility. If attenuation of background noise (like cart machine motors or studio noise) is required, use the *noise gate*: When the input falls below a preset threshold, the output level will be attenuated up to 25dB, depending on the setting of the DEPTH control. If the ambient noise level is relatively low, use the *compressor gate*: When input falls below the preset threshold, compressor gain reduction freezes, then slowly moves to –10dB to prevent annoying noise rush-ups between words, syllables, or instrumental passages.

Built-in **three-band parametric equalization** helps you to achieve a unique, "customized" sound for each voice or instrument, and to compensate for microphone or room characteristics. The center frequency, bandwidth, and boost/cut of each band are adjustable. Tune the high and middle bands for more presence and brightness, or adjust the low band for increased bottom and richness. Digital and LED displays show the exact magnitude of each parameter. All settings are easily modified with the unit's programming keys.

CENTER FREQUENCY:	Low Band — 30–340Hz Mid Band — 210–7,650Hz High Band — 420–15,300Hz
BANDWIDTH:	0.1–5 octaves
BOOST/CUT:	+16dB/–30dB

Excessive sibilance is not only irritating — it can also cause distortion on tape or misbehavior of subsequent audio processors. **Control sibilance** with the Programmable Mic Processor's integral de-esser: a limiter whose sidechain is tuned to sibilance frequencies. This reduces sibilance effectively without adversely affecting high-frequency brightness. The de-esser follows the compressor and parametric equalizer and thus detects and corrects any build-up of sibilance in the processing. A THRESHOLD control determines the amount of de-essing.

Finally, a useful EFFECTS RETURN LEVEL control makes possible **programmable integration of external devices,** such as reverb or psychoacoustic exciters. External processing can be easily inserted even if your console does not have send-and-return capability.

Optional MIDI interface is available to enable you to send and receive program change commands externally. Provision has also been made for control of a separate slave unit for dual-mono or stereo operation.

Consistent Results Every Day

Hours can be wasted trying to recreate successful effects or to reset controls to get a voice or instrument to sound just the way it did before. With its ability to store up to 32 different and complete setups, the same sound is just a few keystrokes away. No more guessing at yesterday's settings, no more scrambling to patch three or four separate processors.

Programming is simple: just select a parameter, enter the desired setting, then move on to the next parameter. Digital and LED displays guide you as you program your setup. When you've set all the controls the way you want them, press STORE to save the settings in memory. From then on, just press RECALL and the program number to instantly reset the controls according to your setup program. Or press COMPARE for quick A/B comparisons of different control set-ups.

The Programmable Mic Processor is designed to protect your work. You can enter a security code to prevent unauthorized tampering with programmed setups. An internal lithium battery helps protect memory from power failures for up to five years.

Applications

Broadcast Mic Processing: Nearly any voice can be made more appealing by the judicious use of mic processing. Each voice at your station can be assigned a permanent setup for each mic/studio combination for instant recall just before the voice airs. The Programmable Mic Processor gives you an important competitive edge by enabling you to get the sound you want *quickly* and *consistently.*

It can also dramatically improve the quality of phone line feeds for talk shows and for remotes by making levels more consistent, improving frequency response, and reducing noise through gating.

Multi-Track Recording and Commercial Production: The 787A can increase efficiency and result in significant savings of time in recording and commercial production facilities. If you often work with the same voice talent, you can conveniently store their "sound" in memory and quickly achieve satisfying results each time you record. Set-ups for instruments can also be reproduced quickly and efficiently. Less time will be spent on tuning processors, so more time can be devoted getting better creative results.

The producer who moves from studio to studio knows that the final product can vary tremendously. The Programmable Mic Processor eliminates some of the uncertainty and variation, enabling you to get closer to your "sound" — regardless of where you are working! The professional artist can note the settings and the particular mic used for an especially satisfying session, and then use these as a reasonable starting point for processing when working in other studios or at live venues.

Installed Sound Systems and Live PA: Why use daisy-chained signal processing when one unit will do? The Programmable Mic Processor takes up less rack space, guarantees that headroom will be used optimally to minimize noise and distortion, and ensures consistent quality. Your talent can concern themselves with their performance, instead of worrying about the sound of the system.

Summary

Add Orban's new Model 787A Programmable Mic Processor to your arsenal. It delivers powerful mic and instrumental processing with an unprecedented combination of integrated, fully programmable functions. Instant access to 32 complete control setups enables you to achieve precise, consistent vocal and instrumental sounds day after day — while saving both time and money.

Estimated First Delivery: Fall '87

Estimated Price: Under $2000

The basic clipper is followed by a 15kHz lowpass filter to assure freedom from aliasing distortion. In addition, difference-frequency distortion below 2.2kHz introduced by clipping is *cancelled* by our patented "Smart Clipper" frequency-dependent distortion-nulling technique. This permits much harder clipping without audible distortion (particularly at high frequencies) than was heretofore possible. The result is dramatically improved high-frequency power-handling capacity.

Frequency Contoured Sidechain (FCS) Overshoot Compensator: While all filter overshoots are minimized by the use of phase correctors (even in the preemphasis network!), the output of the clipper section (including 15kHz lowpass filter and distortion cancellation) contains substantial overshoots due to addition of the distortion-cancellation signal, and to unavoidable overshoots in the lowpass filter. These overshoots must be eliminated without introducing frequency components above 19kHz which would otherwise cause "aliasing" and nonlinear crosstalk in the stereo baseband. The overshoot compensation in the 8000A slightly limits the high frequency power handling capacity of the system. The overshoot compensation system of our major competitor suffers from an inordinate sensitivity to overdrive; it causes *much* more IM distortion than a simple clipper when both are overdriven equally.

We have developed a new "Frequency-Contoured Sidechain" (FCS) overshoot compensator for the 8100A which offers the "best of all possible worlds". It permits high modulation at all frequencies, yet does not suffer from excessive IM (compared to a simple clipper) when overdriven. Simultaneously, it offers extremely good suppression of out-of-band frequency components.

About The Stereo Generator

The stereo generator in the 8100A uses the "matrix" approach to generate the stereo composite signal, as opposed to the more common "switching" approach. It features feedback stabilization of pilot phase, pilot amplitude, *and* separation, optically isolated remote control of stereo/mono function switching, *and* extensive metering.

This high-performance generator features better than 60dB separation (typical), distortion below 0.02%, and superior audio quality because the L+R (dominant in most program material) is not subject to any switching or modulation process, unlike the latest "digital" stereo generators.

Because Part 73.322 of the *FCC Rules* refers to performance requirements for the *stereo generator, exciter, and RF amplifiers*, it is permissible to measure main-to-sub and sub-to-main crosstalk at the input terminals to the stereo generator. The 8100A stereo generator provides internal main-to-sub and sub-to-main crosstalk test modes: a switch permits injecting the output of the right channel processing into either the "main" or "sub" inputs of the stereo generator. An external test box is not required, making verification of crosstalk performance significantly more convenient.

Configuration and Installation

The availability of 25dB of gain reduction range means that the 8100A never needs an external AGC. At stations which install the 8100A at the transmitter and pass audio to the transmitter through an STL with limited signal-to-noise ratio, substantial benefits are achieved by applying compression prior to the STL. For this reason, the compression section of the 8100A can be installed in an Accessory Chassis to permit compression at the studio end.

The Studio Accessory Chassis is low in cost and houses only the dual-band compressor. Its use protects the STL from overload and provides control over most of the processing at the studio. Adjusting the gain between the two chassis is aided by calibrated metering, and requires only an audio oscillator for alignment.

If available, a *composite* STL will allow installation of the entire OPTIMOD-FM at the studio, with the 8100A's baseband connected directly to the STL transmitter.

In either configuration, practical requirements have been carefully considered. Operation is possible with input levels as low as −30dBm, and rigorous RFI shielding is employed. The stereo generator output is held floating over ground to avoid introducing system ground loops when unbalanced exciter inputs are used. An optically-isolated momentary remote switching facility for *stereo, mono-left,* and *mono-right* modes yields maximum versatility and freedom from RFI or ground-loop interference.

To facilitate proofs, a pair of PROOF/OPERATE switches defeat all compression and limiting, yet do not bypass any electronics normally employed in *Operate* mode. To facilitate crosstalk measurements, a NORMAL/MAIN-TO-SUB/SUB-TO-MAIN switch is provided on the stereo generator to establish the necessary test conditions without need for an external test fixture.

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SPECIFICATIONS

The following specifications are presented to satisfy the engineer that they seem reasonable, to help him plan his installation, to help him make certain comparisons with other familiar processing equipment with which he is familiar, to verify that the 8100A can readily pass a Proof of Performance, and to verify that it meets all requirements in the *FCC Rules*.

Years of experience listening to and designing audio processors have convinced us that there are no conventional specifications which correlate well to the listening qualities of a processing system like OPTIMOD-FM. Although it may sound strange to an engineer, we can state with confidence that, given the current state-of-the art in audio measurement techniques, the only way to meaningfully evaluate the distortion introduced by an audio processor is by *subjective listening tests.*

For this reason, no system distortion specifications are presented for the 8100A in *Operate* mode. In this mode, harmonic distortion is highly frequency-dependent, and correlates very well to a listener's actual impression of whether the harmonic distortion test tones emerging from the processor *sound* "distorted" if *listened to* on speakers or phones.

The CCIF Difference-Frequency IM test measures the amplitude of a low-frequency tone produced by the device under test when excited by two high-frequency tones. Because the distortion tone and test tones are far apart in frequency, the distortion tone would not tend to be well-masked by the test tones. Thus CCIF IM is a sensitive indicator of audible distortion—much more so than the more commonly-used SMPTE IM test.

Compared to a simple clipper, the patented distortion-cancelling circuitry in the 8100A "Smart Clipper" is most effective in reducing CCIF IM—thus improving listening quality in a way that does not correlate to more commonly used measurements.

FREQUENCY RESPONSE
(System in *PROOF* mode)
Follows standard 75us preemphasis curve ±0.75dB, 50-15,000 Hz. 50us preemphasis available on special order. All preemphasis networks include a fourth-order lowpass filter and fourth-order phase corrector prior to the high-frequency limiter and clipper to prevent these elements from processing out-of-band program material and to minimize overshoot, thus minimizing the amount of high-frequency limiting and clipping. (NOTE: The Dolby 334 Broadcast Encoder, which is compatible with the 8100A, internally transforms 75us to 25us. Thus broadcasters wishing to use Dolby-B encoding should order standard 75us preemphasis along with the optional Dolby connector.)

INPUT CONDITIONING
Highpass Filter: Third-order Chebychev with 30Hz cutoff and 0.5dB passband ripple. Down 0.5dB at 30Hz; 10.5dB at 20Hz; 31.5dB at 10Hz. Protects against infrasonic destabilization of certain exciters' AFC's, as well as infrasonic gain modulation in the compressor.
Phase Scrambler: Allpass network makes peaks more symmetrical to best utilize the symmetrical peak overload characteristics of the FM medium.

NOISE
−75dB below 100% modulation, 50-15,000 Hz maximum; −81dB typical.

TOTAL SYSTEM DISTORTION
(*PROOF* Mode; 100% Modulation)
Less than 0.05% THD, 50-15,000Hz (0.02% typical); less than 0.05% SMPTE Intermodulation Distortion (60/7000Hz; 4:1).

"MASTER" BAND COMPRESSOR CHARACTERISTICS
Attack Time: approximately 1ms
Release Time: program-controlled—varies according to program dynamics and amount of gain reduction (see text). Process can be scaled fast or slow by means of continuously variable RELEASE TIME control. Employs delayed release for distortion reduction.
Total Harmonic Distortion (measured at VCA output, OPERATE Mode, RELEASE TIME control centered): Less than 0.1%, 50-15,000Hz, 0-25dB gain reduction
Available Gain Reduction: 25dB
Metering: Three dB-linear edgewise-reading gain reduction meters—
MASTER is true peak-reading with electronic acceleration and peak-hold (0-25dB);
COMPRESSOR indicates slow compression component of gain reduction (0-25dB)
LIMITER indicates fast peak limiting component of gain reduction (0-5dB)
Gain Control Element: True VCA. Proprietary Class-A design eliminates crossover notch distortion, modulation noise, and slewrate limiting found in competitive Class-AB designs.

"BASS" BAND COMPRESSOR CHARACTERISTICS
Attack Time: program-controlled; not adjustable.
Release Time: program-controlled; not adjustable. Incorporates delayed-release distortion reduction.
Total Harmonic Distortion (at VCA output, OPERATE mode): Less than 0.1% THD, 50-200Hz, 0-30dB gain reduction.
Available Gain Reduction: 30dB
Metering: single dB-linear edgewise-reading gain reduction meter (0-30dB).
Gain Reduction Element: Proprietary Class-A true VCA
Bass Coupling (U.S. patent #4,249,042): Enables gain of "Bass" band to track gain of "Master" band to any degree, from identical tracking to fully independent operation. Adjustable with BASS COUPLING control.

CROSSOVER CHARACTERISTICS
Control: 6dB/octave @ 200Hz;
Program: 12dB/octave @ 200Hz in unique "distributed crossover" configuration (U.S. patent #4,249,042)

HIGH FREQUENCY LIMITER-CHARACTERISTICS
Attack Time: approximately 5ms
Release Time: approximately 20ms. Delayed release included for distortion reduction.
Mode: Left and right channels operate independently to avoid high frequencies in one channel causing audible timbre modulation of opposite channel.
Control Element: Junction FET
Metering: Two LED's indicate HF limiting in L and R channels.
Threshold of HF Limiting: User-adjustable over 3dB range to meet format requirements

FM "SMART CLIPPER" OUTPUT PROCESSOR CHARACTERISTICS
Nominal Bandwidth: 15.4kHz
Distortion Cancellation: Clipping distortion (below overshoot compensator threshold) cancelled better than 30dB (40dB typical), 20-2200 Hz (U.S. patent #4,208,548).
Delay Correction: Fourth-order allpass
Amount of Clipping: User-adjustable over 6dB range to match format requirements.

FREQUENCY-CONTOURED SIDECHAIN (FCS) OVERSHOOT COMPENSATOR CHARACTERISTICS (patent pending)
System Overshoot: The FCS circuit is best thought of as a "bandlimited safety clipper". It operates like a hard clipper, but does not produce out-of-band frequency components as a simple hard clipper would. Because the audio processing will sometimes limit steady-state material with high average energy (like sinewaves) or with very little high-frequency energy to levels below the threshold of clipping, it is difficult to state a clear and meaningful specification for the *system* overshoot performance of the FCS circuit. The FCS circuit is followed by a safety clipper. The overshoot specification could be slightly improved if this safety clipper were set up to clip more frequently. However, the system is aligned at the factory such that the safety clipper is almost *never* active, thus fully preserving the bandlimiting provided by the FCS circuit. With this safety clipper alignment, the peak modulation will be controlled ±3.5% on arbitrary waveforms clipped to any degree by the FCS circuit (acting as a bandlimited safety clipper); peak modulation will not exceed this level on other material. With typical program material, peak modulation uncertainty is less than 2%.
Sinewave Modulation Ability: 93% modulation (i.e., 0.6dB below maximum overshoot level) at all sinewave frequencies, assuming sinewaves are applied to FCS input.
Dynamic Separation: better than 45dB
Difference-Frequency Intermodulation: FCS circuit causes no more audible IM (such as sibilance splatter) than would a simple hard clipper clipping to the same depth. The entire 8100A processing system is specifically configured to prevent the FCS circuit from audibly degrading the difference-frequency distortion-cancellation properties of the earlier FM "Smart Clipper".

SYSTEM SEPARATION
Greater than 45dB, 50-15,000Hz; 60dB typical

STEREO GENERATOR CHARACTERISTICS
Crosstalk (Main Channel-to-Subchannel, or Subchannel-to-Main Channel): better than −40dB, 50-15,000Hz as measured at input terminals to stereo generator, or using internal crosstalk test mode which applies right-channel audio to either *main* or *sub* stereo generator inputs. Crosstalk representing distortion components (non-linear crosstalk) typically better than −80dB as measured on a baseband spectrum analyzer.
38kHz Subcarrier Suppression: Greater than 40dB below 100% modulation; 60dB typical
Suppression of 76kHz and its Sidebands: Greater than 70dB below 100% modulation
Pilot Frequency: 19.000kHz ±2Hz
Pilot Injection Adjustment Range: Less than 8% to greater than 10% modulation

INPUT
Impedance: greater than 10K ohms, electronically balanced by means of true instrumentation amplifier. Requires balanced source.
Common Mode Rejection: Greater than 60dB @ 60Hz
Sensitivity: −10dBm produces 10dB "Master" Band gain reduction @ 1kHz. Removal of internal 20dB pad permits −30dBm to produce same effect.
Connector: Cinch-Jones 140-style barrier strip (#5 screw).

COMPOSITE (BASEBAND) OUTPUT
Source **Impedance:** 470 ohms, independent of OUTPUT ATTEN setting, unbalanced.
Level: variable 0 to greater than 4V p-p by means of 15-turn OUTPUT ATTEN control
Connector: Type BNC held floating over chassis ground to permit interface to various exciters without need for wideband transformer for ground loop suppression. RF suppressed.
Recommended Maximum Cable Length: 6 ft (1.8m) RG-58A/U

AUXILIARY INPUT/OUTPUT (for Test use only)
Provides L and R lowpass filter output or L and R stereo generator input depending upon setting of rear-apron NORMAL/TEST switch. Connectors are RCA phono-type, unbalanced. Stereo generator requires approx. 3V RMS for 100% modulation, unbalanced, with source impedance of test generator less than 50 ohms.

OPERATING CONTROLS
VU Meter Selector: switches VU meter to read:
L or R Input Buffer
L or R Compressor Out
L or R Filter Out
L-R Level
19kHz Oscillator Level
38kHz PLL Control Voltage
38kHz AGC Control Voltage
±15 V Power Supply Voltages
Stereo/Mono Mode Switch: Momentary front panel switch may be conveniently strapped for either *left* or *right* mono by means of a plug-in internal jumper. Mode may be remote-controlled by application of 6-24 V AC or DC pulses to appropriate rear terminals. Terminals are optically isolated, and may be floated ±50 V above ground. Three pairs of remote terminals will select either *left* or *right* audio inputs in *mono* mode, or *stereo*. Another internal jumper selects which of the three modes will be entered on powerup.

SETUP CONTROLS (front-panel, behind lockable swing-down door)
Compressor:
Left and Right Input Attenuators
"Master" Band Release Time
Gate Threshold
Bass Coupling
Clipping
High-Frequency Limiter Threshold
Stereo Generator:
Pilot Injection
Pilot Phase
L-R Gain (Separation)
Pilot ON/OFF Switch
NORMAL/MAIN-TO-SUB/SUB-TO-MAIN Crosstalk Test Switch (see text)
General:
Output Attenuator
PROOF/OPERATE Switches (to defeat gain reduction, HF limiting, clipping, and gating)
Power ON/OFF
115V/230V Selector Switch

POWER REQUIREMENT
115/230VAC, ±15%, 50-60Hz, approx. 19VA. IEC mains connector with detachable 3-wire "U-Ground" power cord supplied. Leakage to chassis less than 0.25mA @ 115VAC; 0.5mA @ 230VAC.
AC is RF-suppressed.

DIMENSIONS
19" (48.3cm)W x 7" (17.8cm)H x 12.5" (31.2cm)D—4 rack units

SHIPPING WEIGHT
35 lbs; 15.9 kg

ENVIRONMENAL
Operating Temperature Range: 0-50 degrees C (32-122 degrees F).
Humidity: 0-95% R.H., non-condensing

All specifications subject to change without notice.

800014-000-03 3/82

The Multiband OPTIMOD-FM®

Performance Highlights

- ☐ Multiband compressor/limiter/stereo generator
- ☐ Multiband *or* wideband operation plus versatile setup controls permit precise "tuning" for different formats
- ☐ Freedom from processing artifacts and distortion
- ☐ Optimum voice/music balances
- ☐ Excellent high frequency power handling for brightest sound
- ☐ State-of-the-art stereo generator
- ☐ Overshoot compensator permits full modulation at all frequencies, yet doesn't increase low frequency IM distortion

Plus

- ☐ Dual-chassis (studio/transmitter) option
- ☐ Excellent stability and unit-to-unit uniformity
- ☐ Plug-in card construction for easy maintenance
- ☐ Built-in crosstalk test generator
- ☐ Quad coupling ability
- ☐ Rigorous RFI shielding
- ☐ True VCA gain control
- ☐ True peak-reading gain reduction meter
- ☐ Low stereo generator output impedance permits longer baseband cable runs
- ☐ Orban-quality construction, documentation, and backup support

What It Will Do For Your Sound

The Orban OPTIMOD-FM Model 8100A is ideal for any format, and is the best-sounding FM processor that Orban knows how to make. As will be explained in more detail below, it can be operated in either "wideband" or "multiband" modes. Its advantages will be particularly appreciated in formats that ordinarily use heavy processing and that play recently-recorded program material with large amounts of high- and low-frequency energy. Such stations will find that 8100A "heavy processing" is free from the pumping, gain modulation, distortion, and fatigue that many have associated with such processing in the past.

Stations that prefer lighter processing, but that play music with high transient content (such as rock, soul, disco/dance, or jazz), will find that the 8100A (adjusted for slow release times and "multiband" operation) permits use of much more compression than might be expected. Experience with other processors leads many to associate large amounts of compression with highly audible processing—the 8100A will shatter this preconception! And, whether processing is heavy or light, the super-sophisticated release-time circuitry requires much less accurate D.J. gain riding than virtually any other competitive processing.

Operated in "wideband" mode, the 8100A sounds similar to our previous 8000A in formats such as "beautiful music" that ordinarily use light processing and play relatively undemanding program material. The only difference is that processing is even smoother, and no high frequency loss is *ever* apparent on such material. Given the acceptance of the 8000A in such formats, this "family resemblance" assures that use of the 8100A will yield an even more non-fatiguing, audience-pleasing sound—a sound that yields high quarter-hour maintenance and which has resulted in outstanding ratings for so many 8000A "beautiful music" stations.

We are extremely gratified by the 8000A's acceptance among FM broadcasters, and have tried in this design to preserve the 8000A features which accounted for its popularity. We feel that these features include *simplicity of concept, relatively foolproof installation, loud, clean, non-fatiguing sound*, and a "feature" which has little to do with the product per se: a long-term company commitment to quality, reliability, and customer service.

At the same time, we have tried to respond to the demands of the current marketplace to produce a processor which can, when adjusted to do so, produce a "highly-processed" sound which is free from the usual compromises.

The result is a true second-generation overshoot-controlling FM processor. It is at once refined, sophisticated, easy to use, and extremely cost-effective. We believe that, given its versatility and superb sound, it is clearly the best processing investment you can make—and will enhance your quality and ratings now, and in years to come.

We look forward to your giving us the opportunity to show you what the new OPTIMOD-FM Model 8100A can do for *your* station or group.

The following sections describe in some detail the thinking and improvements that went into the new OPTIMOD-FM. We hope that you'll find this information interesting and useful in helping you make decisions about the kind of processing you'll need to stay competitive in the '80's. And we hope to show you how Orban can help you to achieve the ultimate competitive edge.

About the Wideband/Multiband Compressor

Many variations of the traditional parallel-band triband processor have appeared. In our opinion, they all fail critical listening tests because the high frequency band can increase treble uncontrollably, resulting in a phony-sounding high end. Therefore, the 8100A employs a triband structure with a significant difference. The basic compressor consists of two bands—"Master" (above 200Hz), and "Bass" (below). After those two bands are summed together, the combined signal is preemphasized to follow the FM preemphasis curve, and *then* fed into a high-frequency limiter. Unlike the third band of a typical triband compressor, this limiter is very fast, and works only as hard as necessary to avoid audible distortion in the "FM Smart Clipper" following it (discussed below).

Because the "Smart Clipper" can clip far harder than the 8000A could without audible distortion, much less high-frequency limiting is required—and a new level of brightness and high-frequency power handling capability is achieved. The limitations of the 75us preemphasis curve are almost never apparent—even on very bright program material. In the 8100A, the brass retains its "bite"; the cymbals "sizzle". And the "phony highs" phenomenon is entirely absent.

Our dual-band compressor is unique in that it can be operated discriminately (with the bands independent of each other), or wideband.

The "independent" approach is most appropriate for "Top-40", "AOR", "Black", "Disco/Dance", and other formats emphasizing music containing heavy bass and dominant transient material. Program Directors programming these formats usually demand high loudness, high density, and considerable compression. The "independent" approach is ideally suited to these requirements because the requisite midrange density can be achieved without compromising bass definition and punch.

On the other hand, "Beautiful Music", "Fine Arts/Classical", and even some "MOR" stations cannot tolerate the frequency imbalance resulting from the "independent" approach. They often wish to use small amounts of compression and little or no density augmentation to preserve musical values and to avoid long-term listener fatigue. If the music or other program material typical of these formats contains passages without bass, then rumble and noise can be pumped up to an objectionable extent as the "Bass" band attempts to "fill in" with bass that doesn't exist except as noise.

To fill the needs of such formats, a patented BASS COUPLING control permits the "Bass" band to track the "Master" band at all times, except when extremely heavy bass appears at the input. Rather than causing audible gain modulation (as in a *true* wideband system), such bass momentarily causes the gain of the "Bass" band to fall below the gain of the "Master" band. Thus wideband-mode users have the best of both conventional wideband and "independent" approaches. With ordinary program material, the system operates wideband and frequency balances are preserved. If strong bass comes along, the system momentarily becomes "independent" to avoid wideband modulation effects.

The bass coupling is continuously variable between fully independent and fully wideband (retaining excess-bass control) to suit the needs of your market and tastes.

Regardless of the amount of bass coupling, exceptional smoothness is assured by "Smart" attack- and release-time characteristics in all bands. To complement the sophistication of the release-time characteristic, available gain reduction is 25dB in the "Master" band and 30dB in the "Bass" band—an external compressor is *never* required. To make this range usable without "breathing" or "noise rush-up" during pauses, the compressor is gated and "freezes" its gain when the input drops below an adjustable threshold.

The result is an amazingly natural quality—the sort of sound that can hold a listener quarter-hour after quarter-hour—combined with high loudness, superbly appropriate balances between voice and music, a singular absence of perceived distortion, and a new zenith in brightness and high-frequency power handling capability.

About The FM "Smart Clipper"

Loudness is principally dependent upon the design of the peak limiter, lowpass filter, and overshoot compensator. A delicate balance must be maintained between peak/average ratios attained, perceived distortion, and integrity of the baseband spectrum. The 8100A uses a newly designed peak limiting system in which peak limiting, bandwidth limitation, and overshoot control are all elegantly interrelated.

Unlike the questionable "composite clipper" approach which uses brute-force clipping of the stereo composite baseband signal (with attendant aliasing distortion and compromises in dynamic separation), the overshoot compensation schemes used in all OPTIMOD-FM's (including the original 8000A) maintain total integrity of the baseband spectrum. The processing never introduces nonlinear crosstalk, pilot modulation, or other problems inherent in simpler approaches.

Peak Limiting and Distortion Cancellation: To preserve the naturaleness achieved in the multiband compressor, peak limiting is performed by a clipper whose transfer curve has been carefully adjusted to yield minimum perceived distortion in conjunction with the balance of the entire 8100A system. Special circuitry throughout the multiband compressor section assures that the output of the compressor can be applied directly to the clipper—no broadband gain reduction is needed for distortion control. Voice is clean—yet music is loud. And voice/music balances are ideal.

Technical Description

The 672A Equalizer consists of a balanced input buffer amplifier, eight main equalization amplifiers connected in series, and tunable lowpass and highpass positive feedback 12dB/octave Butterworth filters. The output of the lowpass filter is buffered to drive 600 ohms, and is available separately. By suitable switch settings, the main output can be made to carry a highpassed signal. Thus the 672A can be used as an equalizer cascaded with a full electronic crossover, or as an equalizer cascaded with lowpass and highpass filters.

Each amplifier in the equalizer section provides equalization for one band only, thus assuring no interaction between bands. The total equalization is simply the sum (in dB) of the equalizations provided by each of the sections.

Peak boost is accomplished by adding the output of a two-pole bandpass resonator to the main signal; reciprocal dip occurs when this resonator is symmetrically connected as a feedback element in the main equalizer amplifier.

The EQ IN/OUT switch bypasses the last seven main amplifiers and defeats equalization in the first amplifier. Gain and signal polarity are equal in the IN and OUT modes. As the BANDWIDTH control is operated, the skirts of the equalization curve move in and out, but the peak gain and peak frequency remain constant. As the EQUALIZATION controls are operated, the frequencies of peak gain remain constant. However, as the TUNING control or EQUALIZATION control (in *dip* mode) are operated, the bandwidth ("Q") will change, because of the simplifications in the "quasi-parametric" bandpass resonator. Careful design has enabled us to produce curves (in *boost* mode only) essentially identical to the desirable "constant-Q" curves provided by our 622B *true parametric* equalizer in its boost mode.

The EQUALIZATION controls all produce peaking curves; if shelving curves are desired, they can be approximated by tuning the lowest band to 20Hz and the highest band to 20kHz. The breakpoint of the shelving characteristic is then adjusted with the BANDWIDTH control.

Summary

Many people are now aware of the *power* of parametric equalization: the almost sensual satisfaction of getting the sound really *right*. These same people are also demanding professionals, insisting on inaudible noise and distortion, human engineering, quality "feel", and uncompromising reliability.

Orban is well-known for its line of fine parametric equalizers, like the 622B. Now with the 672A, it brings equalization of the same rigorous quality to applications where it could never before be afforded. The 672A is inexpensive enough to qualify it for serious consideration in applications which would otherwise be given by default to a much less able graphic equalizer.

The 672A rounds out the line of Orban "Professionals' Parametrics." Between the 622B *true parametric* and the 672A *quasi-parametric*, there is an equalizer for virtually every need and budget. The Orban "Professionals' Parametrics" are available at your authorized Orban dealer.

Specifications: 672A Equalizer

All specifications apply when driving 600 ohms or higher impedances. Noise measured on an average-reading meter through a 20-20,000Hz bandpass filter with 18dB/octave Butterworth skirts.

Electrical

Input:
Impedance, Load (each leg): 100K in parallel with 1000pF, electronically balanced
Impedance, Driving: Ideally 600 ohms or less, balanced or unbalanced
Nominal Input Level: Between -10 and +4dBm
Absolute Overload Point: +26dBm

Output:
Impedance, Source: 47 ohms in parallel with 1000pF, unbalanced (Optional transformer balanced 600 ohm outputs)
Impedance, Load: Should be 600 ohms or greater-will not ring into any capacitive load
Nominal Output Level: +4dBm
Max. Output Level Before Clipping: +19dBm, 20-20,000Hz

Frequency Response:
±0.25dB, 20-20,000Hz: EQ controls set at "O" detents

Available Gain:
+12dB; adjustable to — infinity by means of front-panel GAIN control

Slew Rate:
Varies between 6 and 13V/uS depending upon setting of GAIN control; slewing is symmetrical. Internal bandlimiting assures that slew rate limiting will not occur even with the most severe equalization and program material.

Square Wave Response:
Square wave exhibits no spurious ringing at any output level. The only ringing observable is that theoretically associated with any given equalization curve.

Total Harmonic Distortion:
Less than 0.05%, 20-20,000Hz (+18dBm)

SMPTE Intermodulation Distortion:
Less than 0.05% (+18dBm: 60/7000Hz, 4:1)

Noise at Output:
Less than -84dBm (EQ In, filters out, controls centered)

Overload/Noise Ratio:
Better than 113dB for any single bandpass filter, for any settings of TUNING or BANDWIDTH controls.

Equalization Ranges:
±16dB peaking EQ, Reciprocal

Tuning Ranges:
20-60Hz; 40-150Hz; 110-310Hz; 230-750Hz; 480-1900Hz; 1.1-4.5Hz; 2.8-9.0kHz; 5.9-21kHz. Dials calibrated at ISO preferred frequencies.

"Q" Range:
Greater than 0.5 to 10 for any setting of the TUNING control

Low Pass Filter Section:
Tunable in 2 ranges: 200-2000Hz or 2.0-20kHz, 12dB/octave, (2nd-order Butterworth)

High Pass Filter Section:
Tunable in 2 ranges: 20-200Hz or 200-2000Hz, 12dB/octave, (2nd-order Butterworth)

Overload Circuit:
Lamp lights for 200mS if the instantaneous peak output of any amplifier rises to within 1dB of its clipping point.

Circuit Design:
Active RC realized with FET-input opamps. Line driver employs discrete transistor current booster.

Operating Temperature:
0-50°C

Power Requirements:
115/230VAC ±10%; 50/60Hz; 6 watts

Physical

Operating Controls:
EQUALIZATION, TUNING, and BANDWIDTH for each of eight bands. TUNING, RANGE (x1; x10), and FILTER IN/OUT for each filter. EQUALIZATION IN/OUT, POWER ON/OFF, and GAIN for entire equalizer.

Panel:
19" x 5¼" (48.3 x 13.3cm): 3 units

Chassis Depth Behind Panel:
5¼" (13.3cm)

Weight:
Net: 8 lbs. (3.6kg); Shipping: 12 lbs. (5.4kg)

AC Cord:
3-wire U-ground to USA Standard

Connectors:
140 type barrier strip (#5 screw) plus parallel-wired ¼" 3 ckt. phone jacks (Switchcraft 12B or equal). Holes punched for XLR-type connectors (Switchcraft D3F and D3M or equal)

Circuit Ground:
Available on barrier strip; normally jumpered to chassis.

Specifications subject to change without notice.

orban

Orban Associates Inc.
645 Bryant Street
San Francisco, California 94107
(415) 957-1067

800007-000-02
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The Orban 672A Equalizer:

more versatility than a graphic;
more economy than a full parametric.

Performance Highlights:

Graphic EQ Section

- Eight bands, each with TUNING and BANDWIDTH control
- Each band tunes over 3:1 frequency range
- "Q" typically variable between 0.3 and 20 (center TUNING)
- ± 16dB equalization range
- EQ controls are long-throw dust-shielded slidepots for good resolution
- TUNING and BANDWIDTH controls marked with "tics" indicating typical settings
- Narrowband notching capability ideal for sound reinforcement
- Bands totally non-interacting

HP/LP Filter Sections

- Each section continuously tunable over 100:1 range in 2 decades
- Each section independently switchable
- 12dB/octave slopes
- Filters follow graphic section. Separate main/highpass and lowpass outputs allow use as filters or as full electronic crossover

General

- Very low noise and distortion
- High slew rate for minimum TIM (SID)
- Front-panel GAIN control; 12dB gain available
- "Peak-stretching" overload lamp warns of clipping anywhere in equalizer
- Active balanced input; unbalanced outputs. Transformer-balanced outputs optional
- RF suppression on input, output, and power leads
- 115/230V, 50-60Hz transformer is standard
- Industrial-grade parts and construction including socketed IC's
- Highly cost-effective

Introducing the 672A Equalizer

The Orban 672A is a cost-effective, professional, quasi-parametric equalizer with the convenience of graphic-type EQ controls. Wide-range high- and low-pass filters with 12dB/octave Butterworth slopes follow the graphic section for added versatility. The 672A has two outputs, arranged so that these filters can also be used as a full tunable electronic crossover.

The 672A is a fully professional product designed to provide a large measure of versatility, convenience, and quality at a very attractive price. While it meets the requirements of the demanding professional, it is also designed and priced to make it understandable and available to the advanced audiophile.

To make the 672A easy to use in situations where its full versatility isn't needed, "tic" marks have been included on the dial calibrations of the TUNING and BANDWIDTH controls. When these controls are set to the tics, the 672A behaves like a standard octave-band graphic equalizer with the eight bands on ISO frequencies from 63 to 8000Hz.

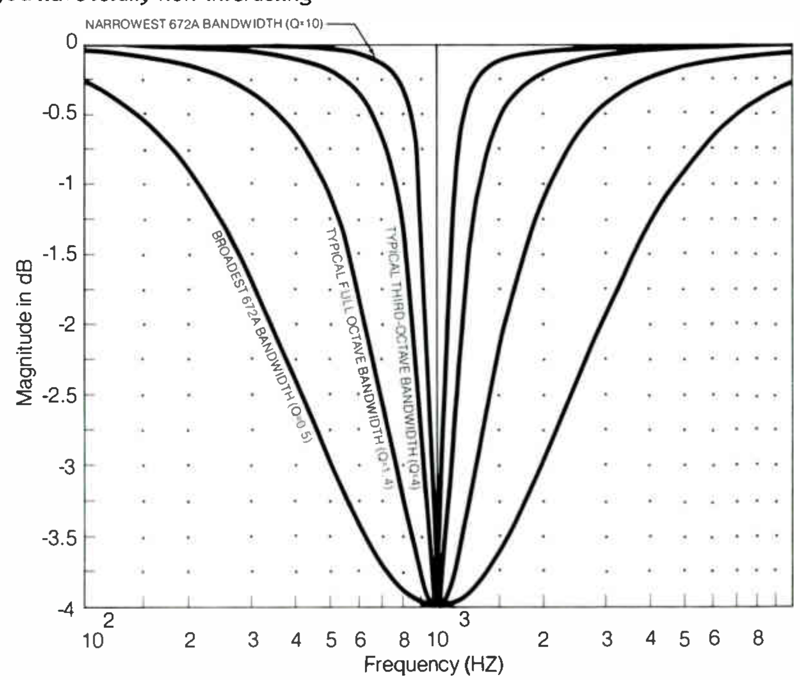
Each feature of the 672A has been thoughtfully chosen and cleverly implemented to make the equalizer a particularly powerful tool in nearly all areas of audio: sound reinforcement, public address, recording studio, broadcasting, motion picture sound, disco, theater . . .

Why "Quasi-Parametric"?

There are two basic types of parametric equalizer: *full-* and *quasi-* parametric. Orban manufactures both types. Both offer far more effective control than other kinds of equalizers, like graphics. Our popular dual-channel *622B* is a *full* parametric. This means that you have *totally non-interacting* control over the three fundamental *parameters* of equalization: the *amount* of peak boost or dip (in dB), the *tuning* (the frequency most affected by the equalization), and the "Q" (which relates to the sharpness of the EQ curve — the degree to which frequencies on either side of the peak frequency are affected by the equalization). As opposed to our *622B* Equalizer, the 672A is *quasi*-parametric. This means that the "Q" changes when you adjust the TUNING and/or EQ controls. Other control adjustments are completely non-interacting: TUNING and EQ do not affect each other.

The other important performance difference between the full-parametric 622B and the new 672A is that the 622B's EQ curves are *"constant-Q"*; the 672A's curves are "reciprocal." "Constant-Q" curves are valuable in that they permit infinite-depth notches to be created; reciprocal curves limit the maximum cut to the same number of dB as the maximum boost. In the case of the 672A, 16dB of cut is available. This is fine for tuning out ring-modes in sound reinforcement systems, but might not be adequate in all circumstances to remove hum or other fixed-frequency interference from a signal. On the other hand, some people prefer reciprocal curves because the boost and cut are mirror images of each other, thus permitting previous equalization to be readily "undone" later. Careful design of the circuitry gives the 672A *in boost mode* a characteristic similar to the 622B's desirable "constant-Q" curve family.

Why did we choose the quasi-parametric technique for the 672A? Because it offers a way to produce a very high quality, stable equalizer at low cost without compromising distortion, noise, accuracy, or reliability.



Various Bandwidth Dipping Curves — 4dB Dip

Applications:

Sound Reinforcement

There are many ways to use the 672A in sound reinforcement:

1) In an economy biamped installation, replace both the third-octave equalizer *and* the electronic crossover with the 672A. The 672A's narrowband, tunable notches can deal with ring modes more effectively than the third-octave unit could. Use three or four of the 672A bands for narrowband notching; leave the rest for wideband EQ.

2) In a higher budget biamped installation, use the 672A as an electronic crossover plus a narrowband, tunable notch filter for ring mode suppression; incorporate a separate third-octave equalizer to correct the house curve.

3) Use variations of (1) and (2) above *with* an electronic crossover; the 672A's high-pass and lowpass filters can then be used to roll off the frequency response of the system in a controlled manner.

4) In a *non-biamped* system (like a stage monitor), use the 672A to equalize the monitor, and use its filters to restrict response in the extreme high and low frequencies.

5) Use the 672A as a *partial* electronic crossover plus equalizer/filter by devoting one channel of 672A equalization to each driver; one filter is required to perform the crossover function; the other can be used for its normal highpass (or lowpass) function.

In all cases, the BANDWIDTH control can be adjusted to make the totally non-interacting (series-connected) bands "combine" — a most desirable characteristic in sound reinforcement.

Any way you cut it, the 672A's economy and extraordinary versatility make it one of the sound reinforcement practioner's most useful tools.

Recording Studios

Every recording studio needs a few channels of 672A equalization to handle the tough chores that the internal console equalizers can't deal with. Patch that problem track through a 672A: its fine-tuning ability lets you clean up the track far more effectively than you could with a graphic or "three knob" console equalizer. Use the tunable filters to help eliminate rumble, cymbal splash, kick drum leakage — you name it!

The 672A is also an ideal adjunct to an electronic music synthesizer — you can create high "Q" formants and shape the spectrum so that the sound comes alive.

If you need to correct the equalization of a finished track because of second thoughts after the mix, the 672A can create the finishing touches as no ordinary equalizer can. It's better than a third-octave graphic, because the 672A can generate broad, non-ringing boosts, whereas the graphic is much more colored and ringy.

Motion Picture Sound

The 672A is an ideal replacement for the graphic equalizers ordinarily used for dialogue equalization in motion picture sound. Set the TUNING and BANDWIDTH controls to the "tics" on the panel, and you get the equivalent of a familiar, easy-to-operate graphic. But when you *need* the extra control and flexibility — such as notching out the extraneous sounds that always seem to plague location recordings — that power is there instantly, without patching or the use of external dip filters. The high and lowpass filters are invaluable for cleaning up noise and rumble without affecting dialogue — and without using up EQ channels to try to achieve filtering. In addition, many "effects" (such as telephone, pocket radio, or "old time" recordings) can be easily created with the 672A alone.

In addition, the 672A can be used to equalize the "B-chain" in the re-recording theater to the acoustic response specifications of the studio. The lowpass filter can effectively simulate the "Academy Rolloff" or its current modifications.

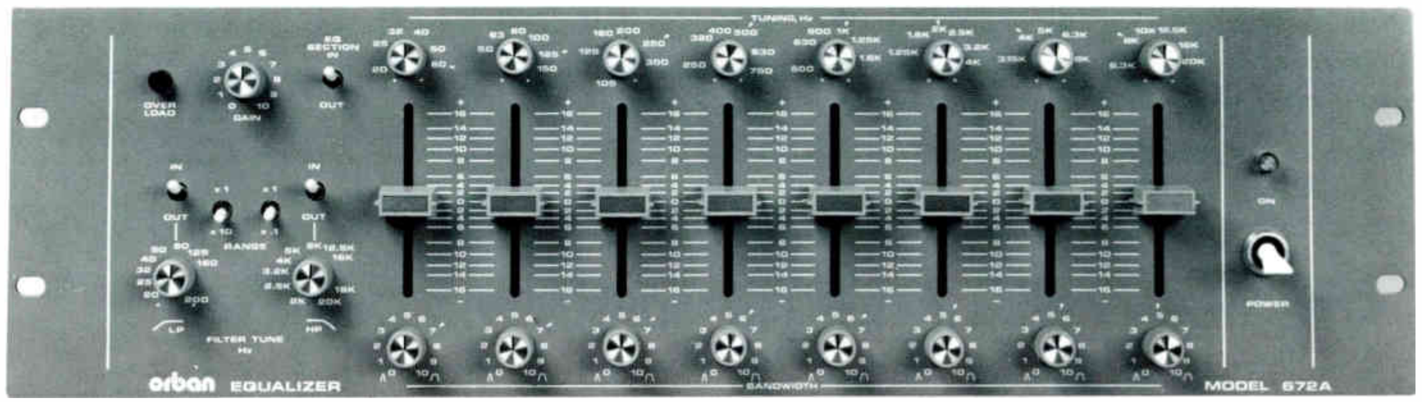
Broadcasting

Use the 672A in the production studio to enhance the announce mike, and to create special production effects that make your station stand out among its competitors. Meanwhile, another 672A can be quietly and efficiently equalizing the program line for maximum punch and brightness on the air. Use the 672A to equalize phone or remote lines for flat response — it's much more versatile than the standard phone company equalizers. In the main studio, use it on the announce mike channel to equalize for maximum presence, and also to notch out sounds like mechanical hum from cart machine motors or air conditioning noise. Whatever your application, the 672A's RF suppression and optional output transformer mean problem-free installation in high-RF environments.

Discos

The 672A is an excellent disco equalizer. The sound contractor installing the system can offer the management exactly the sound desired — including solid, punchy bass free from muddiness and boom — and an aggressive, sizzling top free from the ringing and coloration typical of a full-octave graphic equalizer. The eight bands permit substantial work to be done in flattening out undesired response deviations in the upper bass and midrange. Narrowband notches can even deal with the difficult resonances sometimes encountered in high-efficiency horn-type loudspeakers. In biamped installations, use the separate lowpass and highpass filter outputs as a complete electronic crossover. No other crossover is necessary.

The 672A costs a bit more than an octave-type graphic. But, unlike a graphic, it really *solves* the problems.



OPTIMOD®-TV STEREO GENERATOR

MODEL 8182A/SG: AN ACCESSORY CHASSIS FOR OPTIMOD-TV

HIGHLIGHTS

- ☐ Left and right computer-optimized input filters which significantly exceed BTSC requirements and which essentially eliminate the effect of out-of-band energy on the audio processing.
- ☐ A genuine dbx® encoder card is factory-aligned by dbx to assure tightest conformance with standards.
- ☐ A high-precision realization of the required (and infamous) sum-and-difference 11-pole 15kHz lowpass filters which draws upon every bit of Orban's computer-aided filter design experience to assure the highest standards of accuracy and stability with time and temperature variations.
- ☐ A built-in monitor circuit using a high-precision dbx *decoder* card to simplify verification of stereo generator performance independently of the exciter, RF link, and modulation monitor.
- ☐ A stereo baseband generator based on the proven servo-stabilized matrix design used in our OPTIMOD-FM Model 8100A, literally thousands of which are in service.
- ☐ A user-adjustable baseband clipper to control the large overshoots introduced by the 11-pole filters, preventing IM between the stereo, the SAP, the Pro-channel, and the pilot tone due to exciter overload.
- ☐ Auxiliary input for our separate SAP generator.
- ☐ Prewired card positions for our optional Professional Channel generator.
- ☐ Extensive metering for easy setup and maintenance.
- ☐ Special stereo coder test modes allow alignment of separation and pilot phase using only an oscilloscope.

The 8182A/SAP Generator and the ACC-20 Pro Channel Generator are described in supplements to this brochure. A separate brochure describes the host Model 8182A OPTIMOD-TV Audio Processor.

INTRODUCTION

In the years since the introduction of Orban's original OPTIMOD-FM (Model 8000A), FM stereo has grown up and we've grown with it. There are more Orban FM stereo audio processors and stereo generators in service than all other makes *combined.*

We've brought this extensive experience to the design of the BTSC-standard television stereo *system.* It starts with our refined, field-proven OPTIMOD-TV (Model 8182A) stereo audio processor, which includes the patented CBS Loudness Controller and our exclusive Hilbert-Transform Clipper peak limiter. Our Model 8182A/SG BTSC stereo generator chassis is precisely harmonized to the 8182A to provide the stereo baseband, the Professional Channel, and an input port for the Separate Audio Program (SAP). A full Orban system for BTSC television stereo consists of the following:

- ☐ an 8182A Stereo Audio Processor
- ☐ an 8182A/SG Stereo Generator Accessory Chassis (with Professional Channel option)
- ☐ an 8182A/SAP Second Audio Program Generator (if desired).

The 8182A/SG contains the necessary sharp filtering to protect the pilot and to prevent crosstalk from main channel to subchannel (and vice-versa). A genuine dbx noise reduction encoder card is included to assure most accurate compliance with the BTSC noise reduction standard. The baseband generator is based upon the proven design of the OPTIMOD-FM Model 8100A stereo generator, literally thousands of which are in service. This baseband generator uses a "matrix" approach to generating the baseband, which achieves highest separation (no baseband lowpass filter is necessary) and which subjects the main channel (L + R) to no modulation process which might degrade quality, unlike even the latest "digital" generators.

One unique feature is a built-in calibrated monitor card containing a dbx *decoder* card, along with appropriate deemphasis and dematrixing circuitry. This card permits closed-circuit verification of proper operation of all circuitry but the baseband generator, independent of the RF link and of the modulation monitor. (The performance of the baseband generator can be separately verified by means of several special test modes, conveniently switch-selectable.)

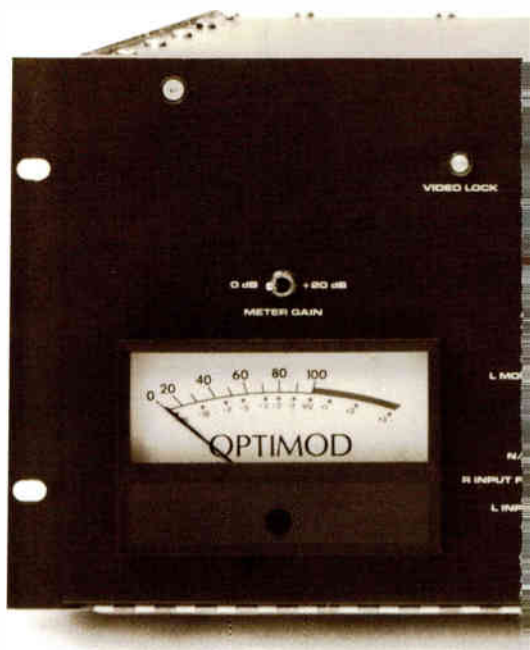
Another special feature is a defeatable baseband clipper with continuously adjustable threshold. This removes the most radical overshoots introduced by the extremely steep filters required by BTSC system recommended practice. (When this clipper is defeated, we have observed short-duration overshoots as large as 300% modulation!)

The mainframe accepts cards to generate the "Professional Channel", which are available as ACC-20.

The mainframe is equipped with an input for a SAP generator. For highest quality, we recommend the Orban 8182A/SAP, which contains audio processing (including Loudness Control) equivalent in quality to that of the 8182A main-channel processor. A dbx encoder card is also included. This product may also be used "stand-alone" without the 8182A or 8182A/SG.

System Configuration: The 8182A/SG is housed in a 7" (4-unit high) chassis which should reside directly above the 8182A. The two chassis are interconnected by means of a cable. The 8182A/SG contains its own power supply: Only audio is passed between the two chassis.

The 8182A/SG is also fully compatible with the Model 8182A/ST Studio Chassis. The 8182A/ST is used to realize the "split configuration",



ACCESS

where the Compressor section of the 8182A resides at the studio end of an STL to protect the STL from overload and to make the most of the operating controls available at the studio. When the 8182A/ST is in use, the main 8182A chassis (containing the high-frequency limiter and peak limiter) and the 8182A/SG Stereo Generator both reside at the transmitter.

Interfacing With OPTIMOD-TV: The 8182A/SG has been specifically designed to mate harmoniously with OPTIMOD-TV Model 8182A to create a system which achieves the highest standards of accuracy and audio quality.

For stations having the older Model 8180A Audio Processor, a factory update is available to make it equivalent to the 8182A. The upgrade adds appropriate wiring plus the Loudness Controller and Hilbert-Transform Clipper features.

Alternately, a field-retrofit kit is available to enable the 8180A to be used with the 8182A/SG without addition of the 8182A's extra features.

The 8182A/SG can be divided into two main sections. The first section performs the L and R lowpass filtering. It is introduced into the signal path of the 8182A processor between the output of the 8182A's Dual-Band Compressor and the input of its High-Frequency Limiter.

The second section, containing the 11-pole sum-and-difference lowpass filters, the noise reduction encoder, and the stereo baseband generator, is connected to the output of the 8182A.

This arrangement permits optimum utilization of the audio processing and results in a system whose performance is significantly better than if the stereo generator were merely cascaded after the audio processor.

Inputs And Outputs: Inputs and outputs are configured to meet any conceivable plant requirement. To avoid ground loops, careful attention has been given to isolating signal grounds from the chassis. Sync may be derived either from the sync bus or from composite video. A loop-through jack is provided. See *Specifications* for levels and impedances.

A SAP input is provided on the 8182A/SG. It accepts the modulated output of an external SAP generator (such as the Orban 8182A/SAP) and sums it with the Stereo and Pro-channel signals to provide the final composite baseband at the 8182A/SG output.

Remote control of STEREO/MONO-LEFT/MONO-RIGHT status, as well as Professional Channel ON/OFF status, is provided by optically-isolated remote control terminals. Another such terminal is provided to remotely display sync lock status.

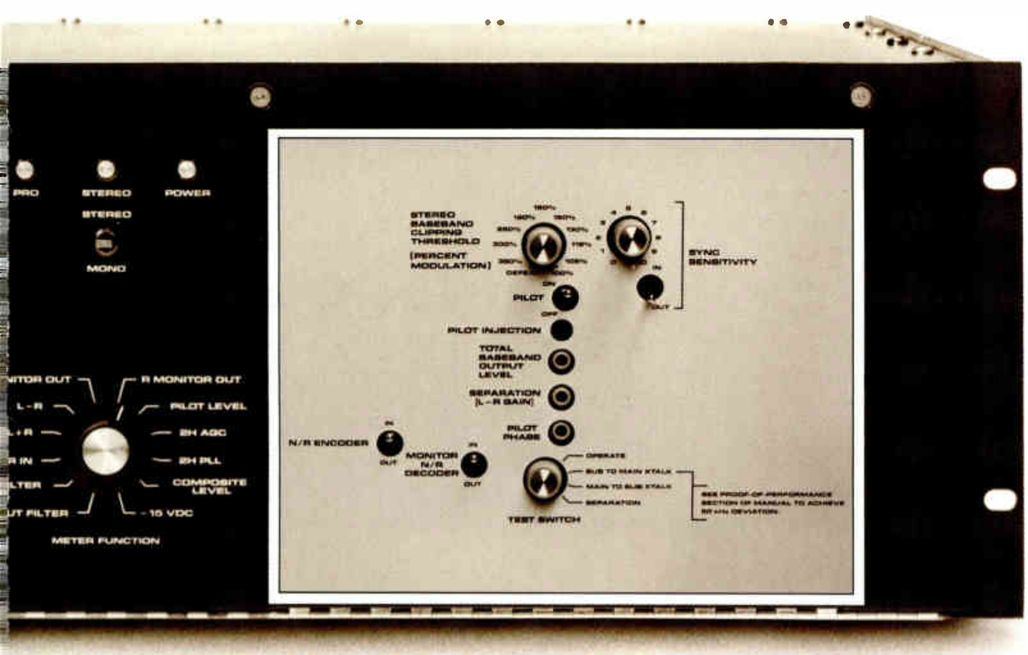
SUMMARY

The Orban 8182A/SG is ideally-matched to the 8182A audio processor. Both units reflect Orban's years of experience in the successful design of audio processing and baseband generating equipment for FM stereo. Together they provide the ideal means of generating BTSC signals of the highest objective and subjective quality.

The circuitry has been subject to intensive computer "sensitivity analysis" to determine the tolerances and temperature coefficients required of individual components. As a result, we are confident that the system will remain stable over time and temperature variations and that each unit will be repairable by simple card-swapping techniques.

The system has been augmented with many features to facilitate easy, accurate setup and maintenance. In addition, Orban provides very complete documentation and customer support.

In short, when experience, reputation, engineering, and systems compatibility are considered, we believe that the OPTIMOD-TV system emerges as the undisputed choice for BTSC stereo transmission. Please contact us at the factory if you have questions, or to obtain the name of your nearest authorized Orban dealer.



DOOR REMOVED SHOWING SETUP CONTROLS

ORDER GUIDE

8182A	OPTIMOD-TV Stereo Audio Processor with CBS Loudness Controller and Hilbert-Transform Clipper.
8182A/SG	BTSC-standard Stereo Generator for use with OPTIMOD-TV Model 8182A.
ACC-20	Plug-in card kit for 8182A/SG to generate Professional subcarrier.
8182A/SAP	Stand-alone BTSC-standard SAP subcarrier generator. Can be used with or without 8182A/SG.

SPECIFICATIONS

(supplementary)

The following specifications refer to a *system* consisting of the 8182A/SG BTSC Stereo Generator and OPTIMOD-TV Model 8182A. These Specifications supplement those for the 8182A. Definitions of various BTSC system terms used below are found in "BTSC System Television Multichannel Sound Recommended Practice", published by the EIA.

FREQUENCY RESPONSE
Sum Channel: Follows standard 75us preemphasis curve ±0.75dB, 50-15,000Hz.
Difference Channel: Follows BTSC standard dbx television noise reduction encoder characteristic with sufficient accuracy to meet system separation specifications (below). The N/R encoder can be switched out and replaced by standard 75us preemphasis to permit measurement of "equivalent stereo separation".

NOISE
−75dB below 100% modulation, 50-15,000Hz maximum; −81dB typical. (Measured with 75us deemphasis in the sum channel and BTSC-standard dbx decoding in the difference channel.)

TOTAL SYSTEM DISTORTION
Less than 0.25% THD, 50-15,000Hz (deemphasized and decoded).

EQUIVALENT STEREO SEPARATION
(no noise reduction)
Better than 40dB, 50-8,000Hz, smoothly declining to better than 26dB at 15kHz. (Exceeds requirements of BTSC Recommended Practice.)

SEPARATION
(10% "75us equivalent-input modulation"; N/R IN)
Better than 30dB, 100-8000Hz, smoothly declining to better than 20dB at 15kHz and better than 26dB at 50Hz. (Exceeds requirements of BTSC Recommended Practice.)

L AND R LOWPASS FILTERS
(see text for location in system)
Type: 6-pole filters with two high-"Q" notches.
Rejection: Better than 50dB at 15734Hz.

L + R AND L − R LOWPASS FILTERS
(see text for location in system)
Type: 11-pole Cauer.
Passband Response: (50-15,000Hz): Typically +0.05, −0.2dB to 15kHz.
Stopband Rejection (15.734kHz and higher): greater than −60dB below Passband Response.

STEREO BASEBAND GENERATOR
Noise And Distortion: substantially below other elements in the 8182A system. The System specifications apply.
Equivalent Stereo Separation (baseband generator *only*): better than 50dB, 50-15,000Hz.
Crosstalk (Main-Channel to Subchannel or Subchannel to Main-Channel; baseband generator *only*): better than −60dB (linear); −80dB (non-linear) referred to ±50kHz deviation.
31.468kHz Subcarrier Suppression: better than 50dB below ±50kHz deviation.
All Other Spurious: better than −65dB below ±50kHz deviation.
Test Modes (see text):
SUB-TO-CROSSTALK
MAIN-TO-CROSSTALK
SEPARATION

INPUTS
Audio: Audio input located on 8182A chassis. See 8182A Specifications.
Sync Lock:
Connectors: Type BNC (2); shell insulated from chassis; looped-through.
Input Impedance: 20,000 ohms; balanced.
Required Level (sync or composite video): 0.6-1.6V p-p; 1V p-p nominal.
Termination: 75 ohms; switchable from rear panel.
SAP:
Connector: Type BNC; shell capacitively coupled to chassis through approximately 500pF for EMI suppression.
Sensitivity: 1.5V pk to produce ±15kHz carrier deviation.
Impedance: greater than 10K, unbalanced but floating over chassis ground.
Professional Subchannel (for optional plug-in Pro Channel card)
Connector: Barrier Strip (#5 screw). EMI-suppressed.
Nominal Input Level: −30 to +8dBm.
Impedance: Greater than 10K balanced bridging.
Remote Control:
(selects MONO LEFT, MONO RIGHT, STEREO modes; also PRO CHANNEL ON/OFF)
Connector: Barrier strip (#5 screw).
Voltage Required For Actuation: 6 to 24V AC or DC, momentary or continuous. 22VDC is supplied on barrier strip to permit use with contact closure.

OUTPUTS
Composite:
Connector: Type BNC, floating over chassis ground.
Level: variable from 0 to greater than 2.2V pk into 75 ohms (for 73kHz total deviation) by means of 18-turn TOTAL BASEBAND OUTPUT LEVEL control.
Output Impedance: 0 or 75 ohms; internally strappable. When strapped for 0 ohms will drive up to 0.047uF in parallel with 75 ohms.
Sync Lock Indicator:
NPN transistor within optoisolator turns ON to indicate successful lock to sync or composite video. Will drive all common logic families with appropriate pullup. Appears on barrier strip (#5 screw).

OPERATING CONTROLS
STEREO/MONO momentary mode switch
VU METER SELECTOR (see **Indicators** below)
METER GAIN switch ((NORMAL/EXPANDED)
Setup Controls (front-panel, behind lockable swing-down door)
- **Sync Lock Card:**
  - SYNC SENSITIVITY
  - SYNC SENSITIVITY CONTROL IN/OUT
- **Stereo Generator:**
  - STEREO BASEBAND CLIPPING THRESHOLD
  - PILOT ON/OFF
  - PILOT INJECTION
  - TOTAL BASEBAND OUTPUT LEVEL
  - SEPARATION (L − R GAIN)
  - PILOT PHASE
  - TEST SWITCH
    - Operate
    - Sub-To-Main Crosstalk
    - Main-To-Sub Crosstalk
    - Separation

INDICATORS
- SYNC LOCK
- PRO CHANNEL ON
- STEREO ON
- POWER ON

VU METER SELECTOR—switches VU meter to read:
- L Filter Out
- R Filter Out
- Noise Reduction In
- Stereo Generator L + R In
- Stereo Generator L − R In
- L Monitor Out
- R Monitor Out
- Pilot Level
- $2F_h$ AGC Control Voltage
- Pilot PLL Control Voltage
- Composite Output (True Peak-Reading)
- −15 Volt Power Supply

POWER REQUIREMENT
115/230VAC, ±15%, 50-60Hz, 35VA. IEC mains connector with detachable 3-wire "U-Ground" power cord supplied. Leakage to chassis is less than 0.5mA. AC is EMI-suppressed.

DIMENSIONS
19″(48.3cm)W × 7″(17.8cm)H × 12.5″(31.8cm)D—4 rack units.

ENVIRONMENTAL
Operating Temperature Range: 0-50°C (32-122°F)
Humidity: 0-95% R.H., non-condensing.

WARRANTY
One year, parts and labor. Subject to limitations set forth in our Standard Warranty.

All specifications subject to change without notice.

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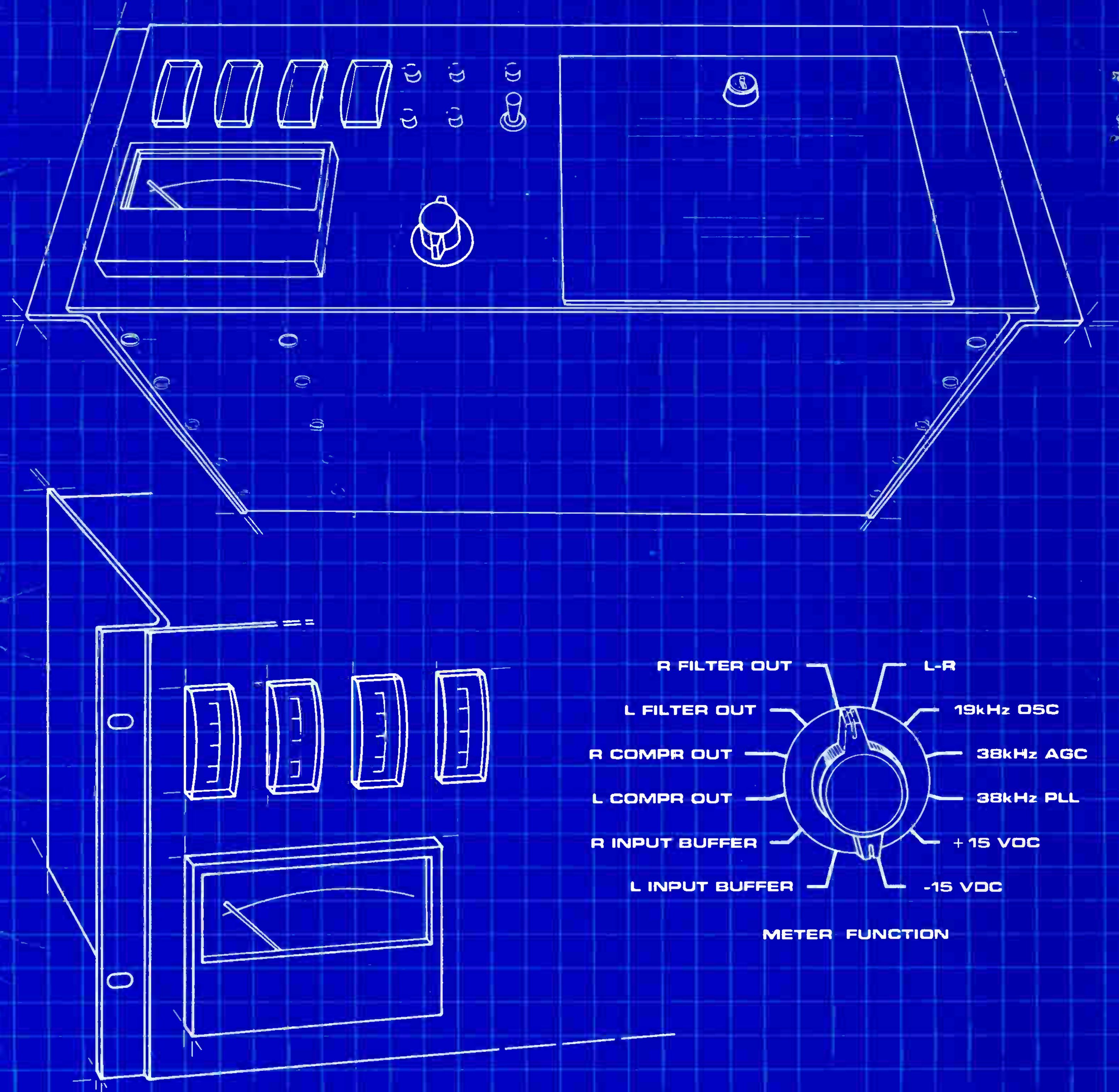


INTRODUCING THE OPTIMOD-FM FOR THE 80's

an applications-oriented technical brochure

INTRODUCING THE ORBAN OPTIMOD-FM MODEL 8100A

Performance Highlights:

- Wideband *or* multiband operation
- More versatile setup controls permit precise "tuning" for different formats.
- New freedom from processing artifacts and distortion
- Improved voice/music balances
- Improved high frequency power handling for brighter sound
- Improved stereo generator with cleaner baseband
- New overshoot compensator permits full modulation at all frequencies, yet doesn't increase low frequency IM distortion

Plus . . .

- Dual chassis (studio/transmitter) option
- Excellent stability and unit-to-unit uniformity
- Plug-in card construction for easy maintenance
- Built-in crosstalk test generator
- Quad coupling ability
- Rigorous RFI shielding
- True VCA gain control
- True peak-reading gain reduction meter
- Lower stereo generator output impedance permits longer baseband cable runs
- Orban quality construction, documentation and backup support

Why A New OPTIMOD-FM?

Our original OPTIMOD-FM, the Model 8000A, is one of the most widely-accepted signal processing devices ever offered to broadcasters. Engineers immediately appreciated the advantages of the 8000A's systems design: the compressor, limiter, high frequency limiter, lowpass filters, and stereo generator are engineered as a single, integrated system and packaged in a single chassis. This patented design permitted us to eliminate loudness-robbing lowpass filter overshoot, and spawned at least half-a-dozen imitators. The system also includes a singularly natural and unstrained compressor/limiter, and an extremely subtle high frequency limiter.

What is probably the most important practical advantage of the 8000A is usually the least appreciated: because of its integrated design, the system is virtually foolproof. Highly critical gain and level relationships between the various system components are factory-set, and cannot be misadjusted in the field. This means that the system can be put into the hands of a RF-oriented engineer who knows little about audio processing, and still produce good results.

The 8000A was designed in 1974, and was "tuned" to the program material available at that time. It's hard to remember now just how different things were then. There was no "disco mix", no "direct-to-disc" fad. AM was still king. Phonograph record equalization practices were much more conservative: today's cartridges can handle much more high and low frequency energy, and producers and engineers are putting it on today's discs.

At the time of the 8000A's introduction, conservatively-processed audio could be at least as loud as the most offensively compressed and limited signals simply by eliminating filter overshoots. The 8000A was thus designed under the assumption that only minimal compression and limiting would be wanted.

Unfortunately, perhaps, for FM listeners concerned about audio quality, overshoot-controlling processors from various manufacturers became widely available, and the loudness war was escalated by combining overshoot control with heavy compression and limiting. Once again, FM quality was being compromised for loudness -- but with overshoot control, everyone was now 3dB louder!

The 8000A's wideband processing is well-suited for moderate gain reduction, slow "open" release times, and conservatively-recorded program material. But it is strained when called upon to create the heavily-processed sound demanded by some of today's program directors. Heavy bass can cause modulation effects (which cannot be eliminated simply by making attack and release time constants "smarter"). The extreme highs on today's records can be audibly dulled by the degree of high frequency limiting necessary to pass them through the 8000A's nonlinear lowpass filter without introducing objectionable distortion. And, if high frequency equalization is added in the production studio, things become even more difficult.

It therefore became clear that new technology and new approaches were needed to achieve the "commercial" sound that many program directors demand — without the offensive side effects heard so often. And, of course, the processor must also be adjustable to achieve a even more open, transparent sound than the 8000A if this is desired by the programming staff. It was also clearly desirable to preserve the many positive attributes of the 8000A -- particularly its essentially "foolproof" nature -- brought about by its integrated systems design and minimum number of setup controls. After four years of careful thinking, experimenting, and listening, the OPTIMOD-FM Model 8100A is finally available to broadcasters.

What It Will Do For Your Sound

The new 8100A is ideal for any format, and is the best-sounding FM processor that Orban knows how to make. As will be explained in more detail below, it can be operated in either "wideband" or "multiband" modes. Its advantages will be particularly appreciated in formats that ordinarily use heavy processing and that play recently-recorded program material with large amounts of high- and low-frequency energy. Such stations will find that 8100A "heavy processing" is free from the pumping, gain modulation, distortion, and fatigue that many have associated with such processing in the past.

Stations that prefer lighter processing, but that play music with high transient content (such as rock, soul, disco/dance, or jazz), will find that the 8100A (adjusted for slow release and "multiband" operation) permits use of much more gain reduction than might be expected. Experience with other processors leads many to associate large amounts of gain reduction with highly audible processing -- the 8100A will shatter this preconception! And, whether processing is heavy or light, the super-sophisticated release-time circuitry requires much less accurate D.J. gain riding than virtually any other competitive processing (including the 8000A).

Operated in "wideband" mode, the 8100A sounds similar to the 8000A in formats such as "beautiful music" that ordinarily use light processing and play relatively undemanding program material. The only difference is that processing is even smoother, and no high frequency loss is <u>ever</u> apparent on such material. Given the acceptance of the 8000A in such formats, this "family resemblance" assures that use of the 8100A will yield an even more non-fatiguing, audience-pleasing sound -- a sound that yields high quarter-hour maintenance and which has resulted in outstanding ratings for so many 8000A "beautiful music" stations.

We have been extremely gratified by the 8000A's acceptance among FM broadcasters, and have tried in our new design to preserve the 8000A features which account for its popularity. We feel that these features include <u>simplicity of concept</u>, <u>relatively foolproof installation</u>, <u>loud</u>, <u>clean, non-fatiguing sound</u>, and a "feature" which has little to do with the product <u>per se</u>: a long-term company commitment to quality, reliability, and customer service.

At the same time, we have tried to respond to the demands of the marketplace to produce a processor which can, when adjusted to do so, produce a "highly-processed" sound which is free from the usual compromises.

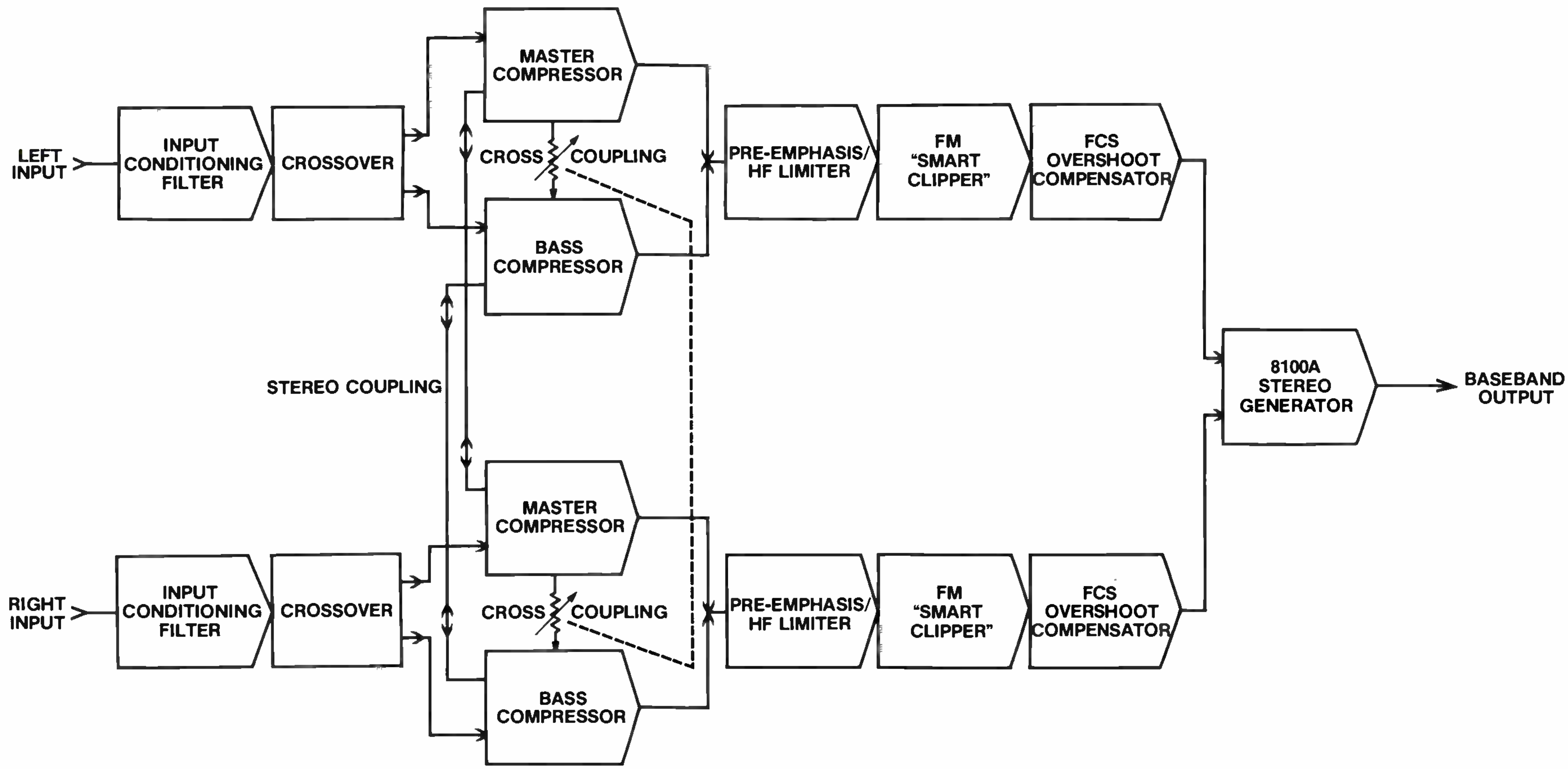
The result is a true second-generation overshoot-controlling FM processor. It is at once refined, sophisticated, easy to use, and extremely cost-effective. We believe that, given its versatility and superb sound, it is clearly the best processing investment you can make -- and will enhance your quality and ratings now, and in years to come.

We look forward to your giving us the opportunity to show you what the new OPTIMOD-FM Model 8100A can do for <u>your</u> station or group.

The following sections describe in some detail the thinking and improvements that went into the new OPTIMOD-FM. We hope that you'll find this information interesting and useful in helping you make decisions about the kind of processing you'll need to stay competitive in the '80's. And we hope to show you how Orban can help you to achieve the ultimate competitive edge.

About The New Wideband/Multiband Compressor

Many broadcasters have tried to increase the loudness and "punch" of their signals by putting triband compressors in front of an 8000A. We have carefully examined such approaches (and have, in fact, experimented with several such designs in the lab). Our conclusion: While such processing can make the signal sound more spectacular on auto or table radios, it does not pass critical listening tests on high-quality component systems. Given the proliferation of such systems (particularly among young, affluent listeners), this is hardly satisfactory!



OPTIMOD-FM MODEL 8100A SIGNAL FLOW DIAGRAM

The major problem is a "phony high end". Naturally subdued program material is falsely brightened -- and record surface noise and/or tape hiss is simultaneously pumped up. Yet when strong high-frequency energy comes along, it is excessively compressed, and its punch, excitement, and impact is removed -- everything loses its integrity! The highs seem subtly detached from the midrange -- as if the musical structure itself were no longer real. The better the monitor system, the more unnatural and unsatisfactory the effect. While such an approach may be valid for AM, FM demands something better.

Our final solution is a triband structure with a significant difference. The basic compressor consists of two bands -- "Master" (above 200Hz), and "Bass" (below). After the two bands are summed together, the combined signal is preemphasized to follow the FM preemphasis curve, and then fed into a high-frequency limiter.

Unlike the third band of a typical triband compressor, this limiter is very fast, and works only as hard as necessary to avoid audible distortion in the "FM Smart Clipper" following it (discussed below).

Because the "Smart Clipper" can clip far harder than the 8000A without audible distortion, much less high-frequency limiting is required -- and a new level of brightness and high-frequency power handling capability is achieved. The limitations of the 75us preemphasis curve are almost never apparent -- even on very bright program material.

Unlike some approaches, the high-frequency limiter cannot increase high-frequency response. While triband "automatic equalization" may work for AM and for non-critical listeners, the audio quality inherent in the FM medium demands individual treatment (preferably in the production studio) of program material judged to have insufficient highs. Most current product, in fact, already has enough high-end energy to challenge the limitations of the 75us FM preemphasis curve -- and the 8100A system reproduces far more of this high end on the air than do typical triband systems. In the 8100A, the brass retains its "bite"; the cymbals "sizzle". And the "phony highs" phenomenon is entirely absent.

Our dual-band compressor is unique in that it can be operated discriminately (with the bands independent of each other), or wideband. The "independent" approach is most appropriate for "Top-40", "AOR", "Black", "Disco/Dance", and other formats emphasizing music containing heavy bass and dominant transient material. Program Directors programming these formats usually demand high loudness, high density, and considerable compression. The "independent" approach is ideally suited to these requirements.

On the other hand, "Beautiful Music", "Fine Arts/Classical", and even some "MOR" stations cannot tolerate the frequency imbalance resulting from the "independent" approach. They usually wish to use small amounts of compression and little or no density augmentation to preserve musical values and to avoid long-term listener fatigue. If the music or other program material typical of these formats contains passages without bass, then rumble and noise can be pumped up to an objectionable extent as the "Bass" band attempts to "fill in" with bass that doesn't exist except as noise. The result, at best, can be an unnatural "heaviness".

To fill the needs of such formats, the two bands can be crosscoupled (patent pending) such that the "Bass" band tracks the "Master" band at all times, except when extremely heavy bass appears at the input. Rather than causing audible gain modulation (as in a true wideband system), such bass momentarily causes the gain of the "Bass" band to fall below the gain of the "Master" band. Thus wideband-mode users have the best of both conventional wideband and "independent" approaches. With ordinary program material, the system operates wideband and frequency balances are preserved. If strong bass comes along, the system momentarily becomes "independent" to avoid wideband modulation effects. In fact, the crosscoupling is continuously variable between fully independent and fully wideband (with excess-bass control), and an infinite number of intermediate settings are possible to suit your tastes.

Regardless of operating configuration, exceptional smoothness is assured by "Smart" attack- and release-time characteristics in all bands. "Hole-punching", "pumping", and "shivering" -- all time-constant problems -- have been reduced to inaudibility by varying these time-constants as a function of the program material. The release-time characteristic used introduces a "dynamic soft-knee" characteristic -- as less compression is used, the program becomes less dense. This preserves an illusion of dynamic range without need for loudness-robbing low static compression ratios. In addition, a "floating platform" release characteristic is introduced as gain reduction exceeds 10dB. The result: a very large amount of gain reduction doesn't make the processed audio sound unnaturally dense.

To complement the sophistication of the release-time characteristic, the available gain reduction has been increased to 25dB in the "Master" band and to 30dB in the "Bass" band -- an external compressor is never required. To make this range usable without "breathing" or "noise rush-up" during pauses, the compressor is gated and "freezes" its gain when the input drops below an adjustable threshold.

The result is an amazingly natural quality -- the sort of sound that can hold a listener quarter-hour after quarter-hour -- combined with high loudness, superbly appropriate balances between voice and music, a singular absence of perceived distortion, and a new zenith in brightness and high-frequency power handling capability.

About The FM "Smart Clipper"

Despite its importance, the multiband compressor did not present the system's greatest design challenge. Loudness is principally dependent upon the design of the peak limiter, lowpass filters, and overshoot compensator. A delicate balance must be maintained between peak/average ratios attained, perceived distortion, and integrity of the baseband spectrum. The 8100A uses a newly designed peak limiting system in which peak limiting, bandwidth limitation, and overshoot control are all elegantly interrelated. It is truly the ultimate vindication of the "system approach", here taken to new heights of sophistication. No collection of casually interconnected boxes can realize the same advantages, because such boxes operate independently of each other.

Unlike the "composite clipper" approach which uses brute-force clipping of the stereo composite baseband signal (with attendant aliasing distortion and compromises in dynamic separation), the overshoot compensation schemes used in all OPTIMOD-FM's (including the original 8000A) maintain total integrity of the baseband spectrum. The processing never introduces nonlinear crosstalk, pilot modulation, or other problems inherent in simpler approaches.

Peak Limiting and Distortion Cancellation: To preserve the naturalness achieved in the multiband compressor, peak limiting is performed by a clipper whose transfer curve has been carefully adjusted to yield minimum perceived distortion in conjunction with the balance of the entire 8100A system. Special circuitry throughout the multiband compressor section assures that the output of the compressor can be applied directly to the clipper -- no broadband gain reduction is needed for distortion control. Voice is substantially cleaner than in the 8000A -- yet music is loud. And voice/music balances are ideal.

The basic clipper is followed by a 15kHz lowpass filter to assure freedom from aliasing distortion. In addition, difference-frequency distortion below 2.2kHz introduced by clipping is cancelled by our patented "Smart Clipper" frequency-dependent distortion-nulling technique. This permits much harder clipping without audible distortion (particularly at high frequencies) than was possible with the 8000A system. The result is dramatically improved high-frequency power-handling capacity.

Because of the complexity of the output section (which could be considered a 28-pole nonlinear filter), we determined that use of active filters would cause excessive sensitivity to component tolerances, drifts, and aging, and would result in reliability insufficient to meet our stringent criteria. We have therefore designed the output section as a "minimum-opamp" realization, and have used passive LC filters wherever practical. These are not "handbook filters". Each has been computer-designed using proprietary Orban programs to fully compensate for the non-ideal characteristics of the inductors, and to achieve filters of extreme accuracy and stability.

Frequency Contoured Sidechain (FCS) Overshoot Compensator: While all filter overshoots are minimized by the use of phase correctors (even in the preemphasis network!), the output of the clipper section (including 15kHz lowpass filter and distortion cancellation) contains substantial overshoots due to addition of the distortion-cancellation signal, and to unavoidable overshoots in the lowpass filter. These overshoots must be eliminated without introducing frequency components above 19kHz which would otherwise cause "aliasing" and nonlinear crosstalk in the stereo baseband. The overshoot compensation in the 8000A slightly limits the high-frequency power handling capacity of the system. The overshoot compensation system of our major competitor suffers from an inordinate sensitivity to overdrive; it causes much more IM distortion than a simple clipper when both are overdriven equally.

We have developed a new overshoot compensator for the 8100A which offers the "best of all possible worlds". It permits high modulation at all frequencies, yet does not suffer from excessive IM (compared to a simple clipper) when overdriven. Simultaneously, it offers extremely good suppression of out-of-band frequency components. We call this new scheme "Frequency-Contoured Sidechain" (FCS) (patent pending).

The next few paragraphs explain the technical details of the FCS system. Things get a bit heavy, so if you're not interested, you should skip to "About the Stereo Generator", below.

Briefly, the FCS system works as follows: The system first derives the overshoots from the input signal. If these overshoots were subtracted from the input signal, the overshoots would be cancelled -- in fact, doing this would be equivalent to simply clipping the input signal. Unfortunately, this can't be done because the overshoots contain out-of-band frequency components which could cause aliasing distortion if applied directly to the stereo generator.

We therefore lowpass-filter the overshoots to eliminate out-of-band components. If the overshoot filter had a flat response to its cutoff frequency, this filtering action would reduce the amplitude of high-frequency overshoots (by removing out-of-band harmonics which make the overshoot "spikey"). This would result in incomplete cancellation of the overshoot after subtraction. The overshoot filter is therefore designed to have a rising frequency response at 15kHz, effectively increasing the gain for the fundamentals of the higher-frequency overshoots and compensating for the fact that their harmonics have been removed. The overshoot extractor and this filter are the "Frequency-Contoured Sidechain".

The overshoot filter has phase shift; phase shift networks are therefore included to make sure that the overshoot subtraction process works correctly, and that the overall FCS system has constant delay.

The rising response of the overshoot filter means that essentially no extra subtraction gain (compared to the system operated without the filter as a simple differential clipper) is required. Any low-frequency IM introduced when the overshoots are derived is therefore no worse than the low-frequency IM caused by a simple clipper.

Because FCS is an instantaneous system and uses no gain reduction or dynamic filtering, it causes neither pumping nor dulling of program material, unlike some less developed overshoot-control systems. If the dual criteria of spectrum control and minimum audible artifacts are used, then the new Orban FCS overshoot compensator stands alone.

About The Stereo Generator

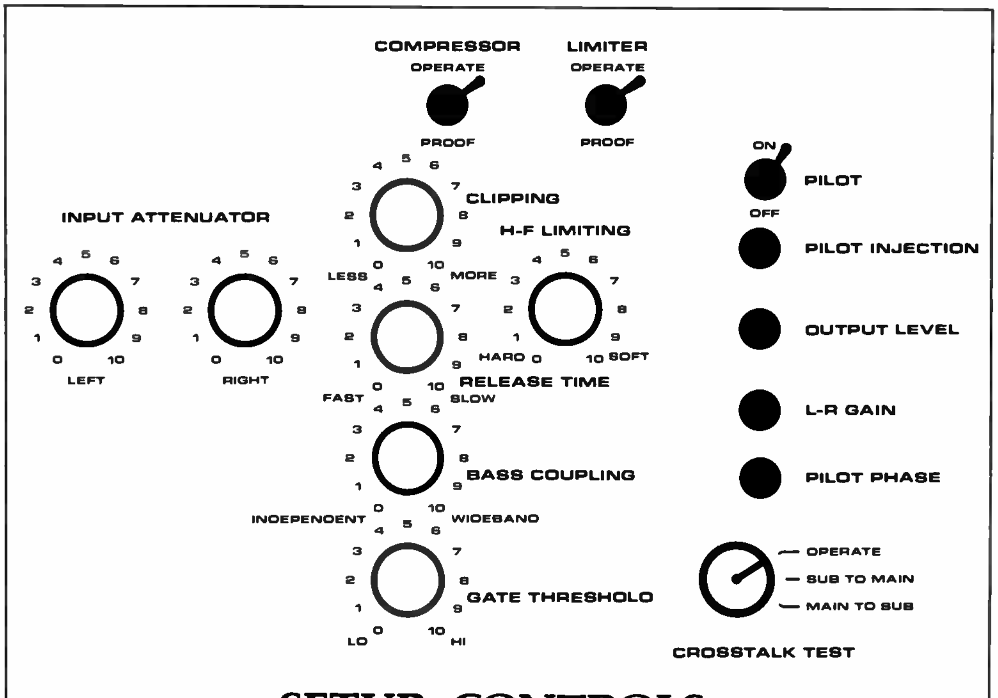
The stereo generator in the 8000A has achieved great respect among engineers because of its stability, excellent separation, low distortion, and clean sound. It uses the "matrix" approach to generating the stereo composite signal, as opposed to the more common "switching" approach.

Research advances and component developments have enabled us to design a refined version for the 8100A. It retains many of the field-proven features of the 8000A stereo generator, including feedback stabilization of pilot phase, pilot amplitude, and separation, optically isolated remote control of stereo/mono function switching, extensive metering, and the use of the "matrix" approach. However, some creative circuit design and the availability of advanced new IC's have enabled us to achieve a better-performing and more elegant design. A newly-designed L-R balanced modulator has increased 15kHz separation to 55dB (typical), and has reduced spurious outputs by 15dB or more (compared to the 8000A) to generate a baseband so clean that transmitted performance is limited entirely by the exciter.

Output impedance has been reduced to 470 ohms (constant at all settings of the OUTPUT ATTEN control) to permit longer runs of coax between OPTIMOD and exciter.

Because Part 73.322 of the FCC Rules refers to performance requirements for the stereo generator, exciter, and RF amplifiers, it is permissible to measure main-to-sub and sub-to-main crosstalk at the input terminals to the stereo generator. In the 8000A, this was done by means of auxiliary test inputs, and required the use of a special test box to create the L+R and L-R test signals. The 8100A stereo generator provides its own main-to-sub and sub-to-main crosstalk test modes: a switch permits injecting the output of the left channel processing into either the "main" or "sub" inputs of the stereo generator. An external test box is not required, making verification of crosstalk performance significantly more convenient.

Differential phase between the left and right channels of the processing is controlled such that the frequency response deviation in the mono sum due to interchannel phase shifts does not exceed 0.2dB -- inaudible to even the most critical listeners.



SETUP CONTROLS

Configuration And Installation

The availability of 25dB of gain reduction range means that the 8100A never needs an external AGC. However, when it is used at the transmitter, unreasonable demands may be placed upon the signal-to-noise ratio of the STL (whether stereo telephone line or dual microwave) if it is called upon to pass unprocessed audio. For this reason, the 8100A is available in two configurations: single-chassis or dual-chassis.

The single-chassis configuration includes the entire system. It is perfect for use with composite STL's, or in installations where studio and transmitter are either on the same site or connected with short, high-quality phone lines.

The dual-chassis version adds a low-cost studio chassis to the system. This chassis houses only the stereo two-band compressor. Driving the STL's, it protects them from overload, and may effectively improve STL signal-to-noise ratio. (The studio chassis accepts three plug-in cards extracted from the main chassis, which are then replaced in the main chassis by two "jumper" cards.)

Adjusting the gain between the two chassis is aided by calibrated metering, and requires only an audio oscillator for alignment.

In both configurations, practical requirements have been carefully considered. Operation is possible with input levels as low as -30dBm, and rigorous RFI shielding is employed. The stereo generator output is held floating over ground to avoid introducing system ground loops when unbalanced exciter inputs are used. And optically-isolated momentary switching of stereo, mono-left, and mono-right modes yields maximum versatility and freedom from RFI or ground-loop interference.

Specifications

Years of experience listening to and designing audio processors have convinced us that there are no conventional specifications which correlate well to the listening qualities of a processing system like OPTIMOD-FM. In fact, designing to achieve very low static harmonic distortion figures almost always results (as with transistor amplifiers) in a deterioration of the listening qualities of the devices! This is because the theoretical basis of harmonic distortion measurements limits their usefulness to indicating small departures from linearity (i.e., distortion) in devices which are supposed to be linear, like amplifiers. A processor is intrinsically non-linear. Its sole purpose in life is to fool the ear into thinking it is linear, while simultaneously lowering peak-to-average ratios and reducing dynamic range in an effort to achieve highest loudness and most efficient utilization of the broadcast channel. Although it may sound strange to an engineer, we can state with confidence that, given the current state-of-the art in audio measurement techniques, the only way to meaningfully evaluate the distortion introduced by an audio processor is by subjective listening tests.

For this reason, no system distortion specifications are presented for the 8100A in Operate mode. In this mode, harmonic distortion is highly frequency-dependent, and correlates very well to a listener's actual impression of whether the harmonic distortion test tones emerging from the processor sound "distorted" if listened to on speakers or phones. For example, odd-order harmonic distortion above 6kHz or so is entirely inaudible because the third harmonic lies above the audible frequency range. Therefore, high-frequency harmonic distortion can be quite high without being perceptible to the ear. Unlike amplifier measurements (where excessive hf harmonic distortion often implies other problems, like TIM), there is no correlation in the 8100A between hf harmonic distortion and other forms of distortion.

Similarly, the SMPTE IM test is almost irrelevant to distortion audibility. The distortion measured by this test is the magnitude of 60Hz sidebands introduced about a 7kHz test signal by the device under test. Because of their frequency proximity to the 7kHz tone, these sidebands are easily "masked" by it -- the test is therefore not sensitive.

On the other hand, the CCIF Difference-Frequency IM test measures the amplitude of a low-frequency tone produced by the device under test when excited by two high-frequency tones. Because the distortion tone and test tones are far apart in frequency, the distortion tone would not tend to be well-masked by the test tones. Thus CCIF IM is a much more sensitive indicator of audible distortion.

Compared to a simple clipper, the patented distortion-cancelling circuitry in the 8100A "Smart Clipper" is most effective in reducing CCIF IM -- thus improving listening quality in a way that does not correlate to more commonly used measurements.

Please note that the 8100A is not an "ultimate loudness box". Operated "fast" and "independent", it can produce about 2dB greater loudness than the 8000A. However exhaustive tests have convinced us that the only way to produce higher loudness than this is to create gross and highly fatiguing processing artifacts and/or distortion which will almost certainly lower ratings in the long run. Indeed, the "point of diminishing returns" has been reached -- when you reach for yet more loudness, things fall apart so rapidly that no one, no matter how crass-sounding, can be significantly louder than an aggressively adjusted 8100A unless they operate outside the FCC Rules.

The following specifications are presented to satisfy the engineer that they seem reasonable, to help him plan his installation, to help him make certain comparisons with other familiar processing equipment with which he is familiar, to verify that the 8100A can readily pass a Proof of Performance, and to verify that it meets all requirements in the FCC Rules. As discussed above, the specifications are not particularly useful in predicting how the 8100A sounds; this must be evaluated by listening tests.

To facilitate proofs, a pair of PROOF/OPERATE switches defeat all compression and limiting, yet do not bypass any electronics normally employed in Operate mode. To facilitate crosstalk measurements, a NORMAL/MAIN-TO-SUB/SUB-TO-MAIN switch is provided on the stereo generator to establish the necessary test conditions without need for an external test fixture.

Frequency Response (System in PROOF mode)

Follows standard 75us preemphasis curve ±0.75dB, 50-15,000 Hz. 50us preemphasis available on special order. All preemphasis networks include a fourth-order lowpass filter and fourth-order phase corrector prior to the high-frequency limiter and clipper to prevent these elements from processing out-of-band program material and to minimize overshoot, thus minimizing the amount of high-frequency limiting and clipping. **NOTE:** The Dolby 334 Broadcast Encoder, which is compatible with the 8100A, internally transforms 75us to 25us. Thus broadcasters wishing to use Dolby-B encoding should order standard 75us preemphasis along with the optional Dolby connector.

Input Conditioning

Highpass Filter: Third-order Chebychev with 30Hz cutoff and 0.5dB passband ripple. Down 0.5dB at 30Hz; 10.5dB at 20Hz; 31.5dB at 10Hz. Protects against infrasonic destabilization of certain exciters' AFC's, as well as infrasonic gain modulation in the compressor.
Phase Scrambler: Allpass network makes peaks more symmetrical to best utilize the symmetrical peak overload characteristics of the FM medium.

Noise

-75dB below 100% modulation, 50-15,000 Hz maximum; -80dB typical.

Total System Distortion (PROOF Mode; 100% Modulation)

Less than 0.05% THD, 50-15,000Hz; less than 0.05% SMPTE Intermodulation Distortion (60/7000Hz; 4:1).

"Master" Band Compressor Characteristics

Attack Time: approximately 1ms
Release Time: program-controlled -- varies according to program dynamics and amount of gain reduction (see text). Process can be scaled fast or slow by means of continuously variable RELEASE TIME control. Employs delayed release for distortion reduction.

Total Harmonic Distortion (measured at VCA output, OPERATE mode, RELEASE TIME control centered): Less than 0.2%, 50-15,000Hz, 0-25dB gain reduction
Available Gain Reduction: 25dB
Metering: Three dB-linear edgewise-reading gain reduction meters --
MASTER is true peak-reading with electronic acceleration and peak-hold (0-25dB);
COMPRESSOR indicates slow compression component of gain reduction (0-25dB)
LIMITER indicates fast peak limiting component of gain reduction (0-5dB)
Gain Control Element: True VCA. Proprietary Class-A design eliminates crossover notch distortion, modulation noise, and slewrate limiting found in competitive Class-AB designs.

"Bass" Band Compressor Characteristics

Attack Time: program-controlled; not adjustable
Release Time: program-controlled; not adjustable. Incorporates delayed-release distortion reduction.
Total Harmonic Distortion (at VCA output, OPERATE mode): Less than 0.2% THD, 50-200Hz, 0-30dB gain reduction
Available Gain Reduction: 30dB
Metering: single dB-linear edgewise-reading gain reduction meter (0-30dB).
Gain Reduction Element: Proprietary Class-A true VCA
Crosscoupling (patent pending): Enables gain of "Bass" band to track gain of "Master" band to any degree, from identical tracking to fully independent operation. Adjustable with BASS CROSSCOUPLING control.

Crossover Characteristics

Control: 6dB/octave @200Hz;
Program: 12dB/octave @200Hz in unique, patent-pending "distributed crossover" configuration

High Frequency Limiter Characteristics

Attack Time: approximately 5ms
Release Time: approximately 20ms. Delayed release included for distortion reduction.
Mode: Left and right channels operate independently to avoid high frequencies in one channel causing audible timbre modulation of opposite channel.
Control Element: Junction FET
Metering: Two LED's indicate hf limiting in L and R channels.
Threshold of HF Limiting: User-adjustable over 3dB range to meet format requirements

FM "Smart Clipper" Output Processor Characteristics

Nominal Bandwidth: 15.4kHz
Distortion Cancellation: Clipping distortion (below overshoot compensator threshold) cancelled better than 30dB (40dB typical), 0-2200 Hz.
Delay Correction: Fourth-order allpass
Amount of Clipping: User-adjustable over 6dB range to match format requirements.

Frequency-Contoured Sidechain Overshoot Compensator Characteristics

System Overshoot: The FCS circuit is best thought of as a "bandlimited safety clipper". It operates like a hard clipper, but does not produce out-of-band frequency components as a simple hard clipper would. Because the audio processing will sometimes limit steady-state material with high average energy (like sinewaves) or with very little high-frequency energy to levels below the threshold of clipping, it is difficult to state a clear and meaningful specification for the system overshoot performance of the FCS circuit.

The FCS circuit is followed by a safety clipper. The overshoot specification could be slightly improved if this safety clipper were set up to clip more frequently. However, the system is aligned at the factory such that the safety clipper is almost never active, thus fully preserving the bandlimiting provided by the FCS circuit. With this safety clipper alignment, the peak modulation will be controlled ±3.5% on arbitrary waveforms clipped to any degree by the FCS circuit (acting as a bandlimited safety clipper); peak modulation will not exceed this level on other material. With typical program material, peak modulation uncertainty is less than 2%.

Sinewave Modulation Ability: 93% modulation (i.e., 0.6dB below maximum overshoot level) at all sinewave frequencies, assuming sinewaves are applied to FCS input.

Dynamic Separation: better than 45dB

Difference-Frequency Intermodulation: FCS circuit causes no more audible IM (such as sibilance splatter) than would a simple hard clipper clipping to the same depth. The entire 8100A processing system is specifically configured to prevent the FCS circuit from audibly degrading the difference-frequency distortion-cancellation properties of the earlier FM "Smart Clipper".

System Separation

Greater than 45dB, 50-15,000Hz; 60dB typical

Stereo Generator Characteristics

Crosstalk (Main Channel-to-Subchannel, or Subchannel-to-Main Channel): better than -40dB, 50-15,000Hz as measured at input terminals to stereo generator, or using internal crosstalk test mode which applies left-channel audio to either main or sub stereo generator inputs. Crosstalk representing distortion components (non-linear crosstalk) typically better than -80dB as measured on a baseband spectrum analyzer.

38kHz Subcarrier Suppression: Greater than 40dB below 100% modulation; 60dB typical

Suppression of 76kHz and its Sidebands: Greater than 70dB below 100% modulation

Pilot Frequency: 19.000kHz ±2Hz

Pilot Injection Adjustment Range: Less than 8% to greater than 10% modulation

Input

Impedance: 600 ohms, electronically balanced by means of true instrumentation amplifier. Requires balanced source.

Common Mode Rejection: Greater than 60dB @60Hz

Sensitivity: -10dBm produces 10dB "Master" Band gain reduction @1kHz. Removal of internal 20dB pad permits -30dBm to produce same effect.

Connector: Cinch-Jones 140-style barrier strip (#5 screw).

Composite (Baseband) Output

Source Impedance: 470 ohms, independent of OUTPUT ATTEN setting, unbalanced.

Level: variable 0 to greater than 4v p-p by means of 15-turn OUTPUT ATTEN control

Connector: Type BNC held floating over chassis ground to permit interface to various exciters without need for wideband transformer for ground loop suppression. RF suppressed.

Recommended Maximum Cable Length: 6ft (1.8m) RG-58A/U

Auxiliary Input/Output (for Test use only)

Provides L and R lowpass filter output or L and R stereo generator input depending upon setting of rear-apron NORMAL/TEST switch. Connectors are RCA phono-type, unbalanced. Stereo generator requires approx. 3V RMS for 100% modulation, unbalanced, with source impedance of test generator less than 50 ohms.

Operating Controls

VU Meter Selector: switches ASA-standard VU meter to read:

L or R Input Level
L or R Compressor Output
L or R Filter Out
L-R Level
19kHz Oscillator Level
38kHz PLL Control Voltage
38kHz AGC Control Voltage
$\pm$15 V Power Supply Voltages

Stereo/Mono Mode Switch: Momentary front panel switch may be conveniently strapped for either left or right mono by means of a plug-in internal jumper. Mode may be remote-controlled by application of 6-24 V AC or DC pulses to appropriate rear terminals. Terminals are optically isolated, and may be floated +50 V above ground. Three pairs of remote terminals will select either left or right audio inputs in mono mode, or stereo. Another internal jumper selects which of the three modes will be entered on powerup.

Setup Controls (front-panel, behind lockable swing-down door)

Compressor:
Left and Right Input Attenuators
"Master" Band Release Time
Gate Threshold
Bass Crosscoupling
Clipping
High-Frequency Limiter Threshold
Stereo Generator:
Pilot Injection
Pilot Phase
L-R Gain (Separation)
Output Attenuator
PROOF/OPERATE Switches (to defeat gain reduction, hf limiting, clipping, and gating)
Pilot ON/OFF Switch
NORMAL/MAIN-TO-SUB/SUB-TO-MAIN Crosstalk Test Switch (see text)
Power ON/OFF
115V/230V Selector Switch

Power Requirement

115/230VAC, $\pm$15%, 50-60Hz, approx. 19VA. IEC mains connector with detachable 3-wire "U-Ground" power cord supplied. Leakage to chassis less than 0.5mA. AC is RF-suppressed.

Dimensions

19"(48.3cm)W x 7"(17.8cm)H x 12.5"(31.2cm)D -- 4 rack units

Environmental

Operating Temperature Range: 0-50 degrees C (32-122 degrees F).
Humidity: 0-95% R.H., non-condensing

Warranty

One year, parts and labor. Subject to limitations set forth in our Standard Warranty.

All specifications subject to change without notice.

orban

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Telex: 17-[illegible]480, Cable: ORBANAUDIO

Technical Description

The 622 Parametric Equalizer consists of a balanced input buffer amplifier, an input attenuator, and four peak/dip equalization sections connected in *series*, assuring no interaction between sections. The final section contains a current booster capable of driving 600 ohm loads. The output of the input buffer and each of the equalization sections is monitored at all times by the overload indicator. The EQ IN/OUT switch bypasses all circuitry but the input buffer and output amplifier; it is arranged so that gain and signal polarity are maintained constant in the IN and OUT modes.

Equalization is accomplished by summing the output of a two-pole bandpass filter to the main signal in-phase (for boost) or out-of-phase (for cut). This creates the "Constant-Q" curves described above.

As the BANDWIDTH control is operated, the skirts of the equalization curve move in and out but the peak gain remains constant (see Fig. 1). As the TUNING control is operated, the curves in Fig. 1 slide along the frequency axis but their shape is unchanged. If shelving characteristics are desired, they may be approximated by adjusting the lowest band to 20Hz and the highest band to 20kHz. The breakpoint of the shelving characteristic is then adjusted with the BANDWIDTH control.

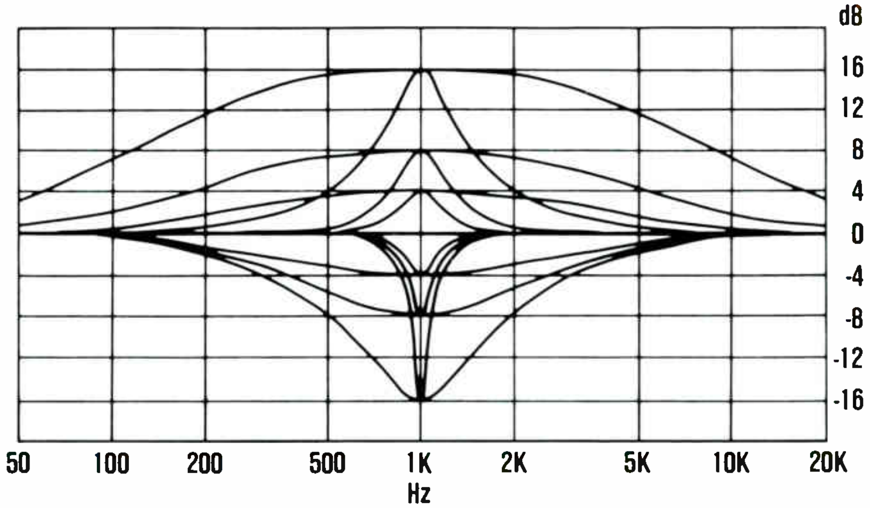

Figure 1

Performance Specifications

Specifications apply to each channel except as noted. All specifications apply when equalizer drives 600 ohm or higher impedances. All noise specifications assume a 20-20,000 Hz bandpass filter with 18 dB/octave Butterworth skirts.

Operating Controls: EQUALIZATION, EQUALIZATION IN/OUT, BANDWIDTH, and TUNING for each of four bands. MASTER EQUALIZATION IN/OUT, GAIN, POWER ON/OFF.
Frequency Response: (EQ controls set mechanically flat) ±0.25 dB, 20-20,000 Hz.
Available Gain: + 12 dB, adjustable to -∞ by means of front-panel GAIN control.
Input: (RF suppressed)
Impedance: (each leg) 100K in parallel with 1000pF, electronically balanced. Driving impedance should be 600 ohms or less.
Absolute Overload Point: + 26dBm.
Output: (RF suppressed)
Level: greater than + 19 dBm into 600 ohms, 20-20,000 Hz
Impedance: 47 ohms in parallel with 1000pF, unbalanced. (Option 01 provides a transformer-balanced output for both channels)
Equalizer is unconditionally stable and will not ring with any captive load.
Risetime: less than 4 microseconds.

Slew Rate: greater than 6 V/microsecond. Internal bandlimiting assures that slew rate limiting will not occur with even the most severe equalization and program material.
Square Wave Response: Square wave exhibits no spurious ringing at any output level. The only ringing observable is that theoretically associated with any given equalization curve.
Circuitry: active RC, utilizing FET-input IC opamps. The output line driver utilizes a discrete transistor current booster.
Total Harmonic Distortion (+ 18 dBm output): less than 0.025%, 20-20,000 Hz. Typically less than 0.002% at 1kHz, + 18 dBm.
SMPTE Intermodulation Distortion: Typically 0.008% at + 18 dBm equivalent peak output, using 60 Hz/7 kHz; 4:1.
Noise: At Output, GAIN control adjusted for unity gain, all EQ switches IN, all EQ controls FLAT: Less than -84 dBm; -87 dBm typical.
Overload-to-noise Ratio of Single Parametric Bandpass Filter: greater than 102 dB for any combination of TUNING and BANDWIDTH settings.
Interchannel Crosstalk, 622B dual-channel equalizer: less than -90 dB, 20-20,000 Hz.

Equalization Characteristics: Figure 1 shows curves corresponding to the maximum and minimum bandwidths for each band. DB equalization contributions of the individual bands add without interaction. BANDWIDTH, TUNING, and EQUALIZATION controls are all continuously variable.
Range of Adjustment of "Q": 0.29 to 3.2.
Range of Adjustment of Peak Equalization: + 16 dB to -∞ . Typical notch depth obtainable is 40 dB.
Tuning Range (per band): 20-500 Hz, 68-1700 Hz, 240-5850 Hz, 800-20,000 Hz. Tuning dials are calibrated at ISO preferred frequencies.
Power Requirements: 115/230 volt 50-60 Hz AC, approximately 4 watts (622A), 7 watts (622B). Captive "U-Ground" power cord. Option 02 eliminates the AC power supply. Power requirements for the Option 02 version are ± 18 to 28 volts DC at 60 ma per equalizer channel. Option 02 is supplied on special order only, and is recommended only for users planning to install a large number of 622 channels in a given installation.
Overload Lamp: will light for approximately 200 mS if the instantaneous peak output of any amplifier in the equalizer is driven within 1 dB of its clipping point.
Size: 19" (48.3 cm) wide x 3.5" (8.9 cm) high x 5.2" (13.3 cm) deep.
Shipping Weight: 10 lbs. (4.5 kg).

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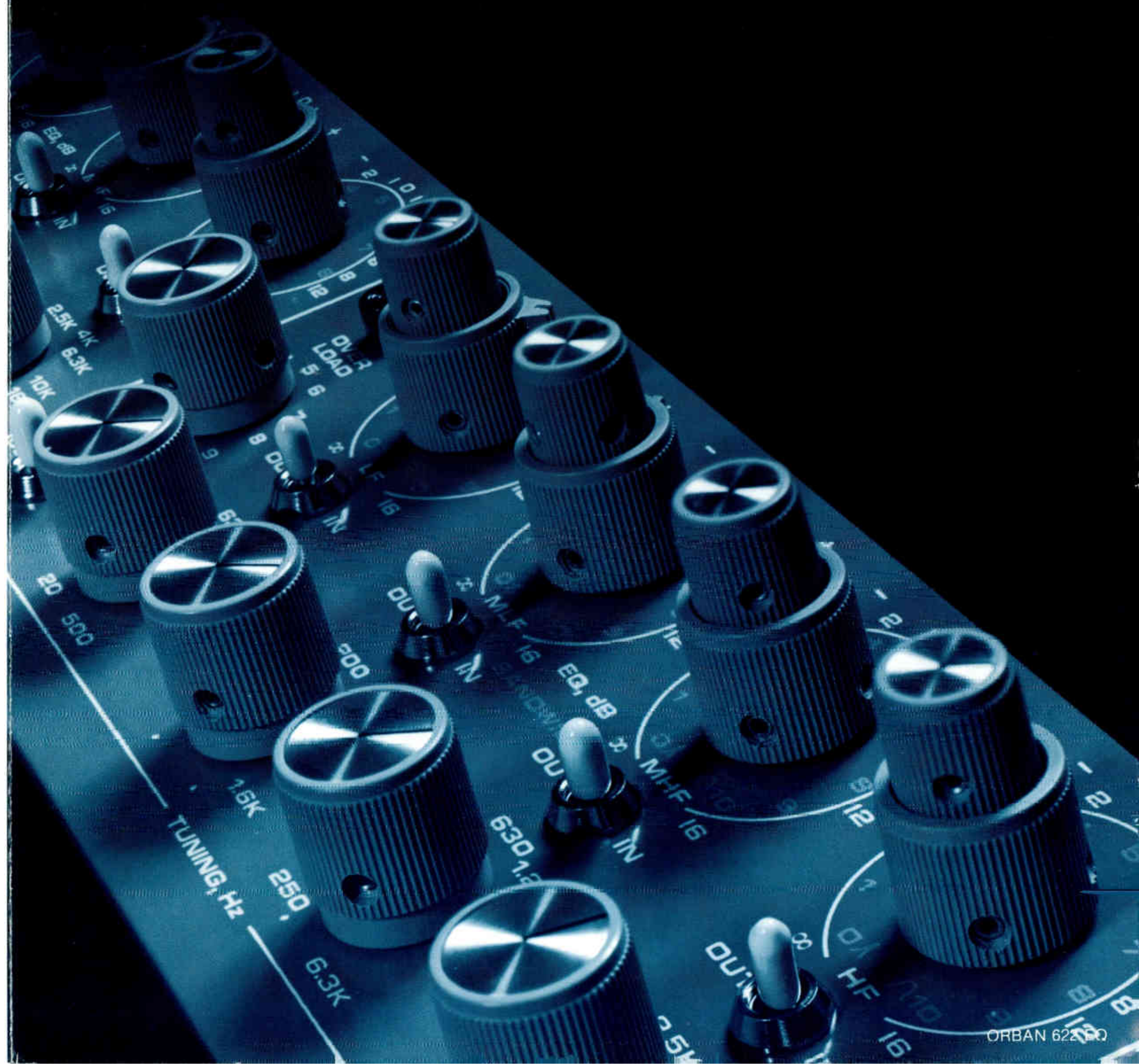
The Orban 622 Parametric Equalizer

The World Class Parametric EQ



Performance Highlights

- +16dB, –infinity dB equalization range
- Each section tunes over 25:1 frequency range
- "Q" adjustable from 0.29 to 3.2
- "Constant-Q" operation enables use of equalizer as notch filter
- True Parametric operation: all controls non-interacting
- Four totally non-interacting peak boost/cut sections, each with TUNING and BANDWIDTH control
- Front panel GAIN control; 12dB gain available
- In/out switches for each section, as well as entire equalizer
- "Peak stretching" overload indicator warns of overload anywhere in equalizer
- Active balanced input; unbalanced output. Transformer-balanced output available
- Very low noise and distortion
- High slew rate for minimum TIM (SID)
- High stability active RC circuitry
- Single or dual-channel models
- RFI suppression of input, output, and power leads
- 115/230V 50-60 Hz AC power supply standard

The 622 Parametric Equalizer

Description
The Orban 622 is a true Parametric Equalizer of high professional quality, providing outstanding versatility and control. The four sections in each channel each use "constant-Q" circuitry. This results in an equalizer of outstanding musicality, and permits any section to be used as a narrowband notch filter (with typically better than 40dB rejection) to effect room tuning or to eliminate fixed-pitch interference, like hum. The sections are totally non-interacting: the total equalization (in dB) is simply the sum of the equalizations of the individual sections.

Considerable attention has been devoted to human engineering, maintainability, and performance in the harsh environments often encountered by the professional user. Levels, impedances, and connectors are fully compatible with virtually all professional equipment. The rugged chassis provides shielding against electrical interference, RFI, and dust. Reliability is assured by a formal burn-in program and additional high temperature burn-in of most semiconductors.

Each feature of the 622 Parametric Equalizer has been thoughtfully chosen and implemented to make the equalizer a particularly powerful tool in nearly all areas of audio: sound reinforcement, public address, recording studio, broadcasting, motion picture sound, disco, theater . . .

Parametric Equalizers—An Explanation
In general, "Parametric" means anything an equalizer manufacturer chooses it to mean. The most commonly accepted definition is that a "Parametric" equalizer provides continuously variable control over the three fundamental *parameters* of equalization: the *amount* of peak or dip (in dB), the *frequency* (the "center frequency") at which the maximum peak (or dip) occurs, and the *bandwidth* (the number of frequencies on either side of the center frequency which are affected by the equalization.)

Bandwidth is a poorly defined parameter for equalizers. (In particular, it is *not*, as often stated, the ratio of the center frequency to the frequency at which the equalization is 3dB down — what if we're using only 2dB eq?!) The bandwidth is related to a more precisely defined factor called the "pole Q", or simply the "Q", for short. If this factor is kept constant as the EQ control is operated, then a curve family called "constant-Q" (see Fig. 1) is created. These curves are not reciprocal —the dip curves are narrower than the boost curves. Experience has shown that these curves produce a more musically-useful equalization with minimum readjustment of the BANDWIDTH control as the EQ control is operated. Moreover, unlike the more common reciprocal curves, they permit the creation of deep narrowband notches which are highly useful for suppressing sounds of fixed pitch, with negligible degradation of the rest of the program.

There are two fundamentally different types of Parametric Equalizer. Orban manufactures both types. The 622 is a "true Parametric." This means that adjustment of a single parameter (like the center frequency) does not affect the other two parameters. This configuration is preferred when maximum convenience is desired. Conversely, a "quasi-Parametric" (like our 672A) permits some interaction (usually changing center frequency also changes "Q") to achieve lower cost. For more detailed information on these important but challenging subjects, please request our free paper "How To Choose Equalizers For Professional Applications" by Robert Orban.

Regardless of configuration, Parametric Equalizers are usually superior to other types when maximum control, flexibility, and freedom from undesired side-effects are desired. In using non-parametric equalizers, you must live with whatever bandwidth and whatever discrete center frequencies the manufacturer has chosen. And you don't have any control over how the bandwidth varies as you change EQ. In graphic equalizers, large boosts over a broad bandwidth often become excessively colored and ringy compared with the results obtainable from an optimally adjusted Parametric.

Applications

Sound Reinforcement
The 622 can often do a surprisingly effective job of "tuning" a sound reinforcement system to a room. The availability of four narrowband notches means that sharp resonances can be dealt with—often more effectively than with third-octave filter sets with fixed filter frequencies. While not designed to replace third-octave filters, the 622 can often augment their effectiveness and in many cases can make surprisingly substantial improvements all by itself. One useful variation is to use both equalizers in the 622B in series—one to notch out feedback and one to provide broadband equalization.

In large scale reinforcement systems for traveling shows, the Orban Parametric is highly useful in equalizing stage monitor systems. In bi-amped and tri-amped installations, the use of one channel of Parametric equalization after each output of the electronic crossover has proven to be of substantial value in optimizing the performance of the individual drivers in the loudspeaker system. Anywhere a conventional equalizer is used, the 622 can do the job better. If more complicated equalization is required, use several channels in cascade. The noise level is low enough to permit this.

Motion Picture Sound
The 622 is an ideal replacement for the graphic equalizer ordinarily used for dialogue equalization. The mixer gets finer control, plus the ability to instantly notch out the extraneous sounds that always seem to plague location recordings. In the music recording studio, the 622's improved adjustability means better sound in the theater. In production, use it for special effects like telephone or "old time" recordings.

Recording Studios
Every recording studio needs at least a few channels of Parametric equalization to handle the tough chores that the internal console equalizers can't deal with. Many experienced Orban Parametric users prefer to have one channel of Parametric equalization on each console input. With practice they're fast and easy to use, and the powerful features (like notching and fine-tuning are instantly available without patching).

The 622 is a particularly valuable adjunct to an electronic music synthesizer—you can create high "Q" formants and shape the spectrum so that the sound comes alive.

If you need to correct the equalization of a finished track because of second thoughts after the mix, the 622 can create the finishing touches as no ordinary equalizer can. It's better than even a third-octave graphic, because the 622 can create broad, non-ringing boosts—the graphic is much more colored and ringy.

Broadcasting
Use the 622 in the production studio to enhance the announce mike and to create special production effects that make your station stand out among its competitors. Meanwhile, another 622 can be quietly and efficiently equalizing the program line for maximum punch and brightness on the air.

Use the 622 to equalize phone and remote lines for flat response—it's much more versatile than the standard phone company equalizers. In the main studio, use it on the announce mike to equalize for maximum presence and also to notch out undesired sounds like mechanical hum from cart machine motors or air conditioning noise. Whatever your application, the 622's RFI suppression and optional balanced-output transformer mean trouble-free installation even in high-RF environments.

In Summary

Many people are now aware of the *power* of parametric equalization: the almost sensual satisfaction of getting the sound exactly *right*. These same people are also demanding professionals, insisting on inaudible noise and distortion, human engineering, quality "feel," and uncompromising reliability. We at Orban feel that there is no cheaper equalizer that delivers this full degree of professionalism, and no more expensive equalizer which provides an improvement in performance proportionate to its cost. Our 622 is also backed up by a company which is firmly established in the industry and is committed to service, stability, and responsiveness to customer needs. That's why our 622 is such a fine choice for any professional who needs an equalizer.

The 622 is available from your local Orban professional audio dealer. Call or write for the name of the dealer nearest you.

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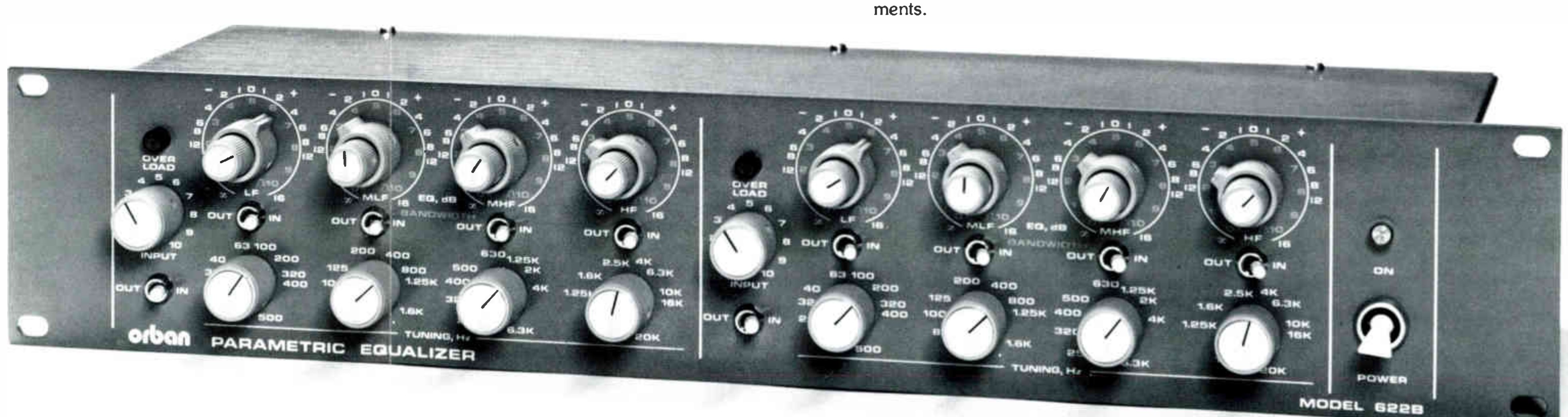

Applications

The 424A can be applied anywhere that current compressors, limiters, and/or de-essers are used, since its versatility does not limit it to a single "sound," but instead lets it operate as a smooth, gentle compressor, a peak limiter, a de-esser, or any combination of these — all with a singular absence of undesirable artifacts.

This means that the unit can be used in recording studios, in broadcast production studios, ahead of broadcast studio-transmitter links, in sound reinforcement, and in video production and sweetening. It is an ideal all-in-one vocal processor, combining the necessary AGC and de-esser functions. It also shines when smoothly handling **mixed** program material — making it excellent for preparing cassette duplicating masters, or protecting tape recorders and cart machines from overload in tight time-pressure situations in broadcast and recording. Simultaneously, its versatile, wide-range setup controls make it a natural for processing single instrumental or vocal tracks in studios. Exploitation of the VCA clipping feature can result in substantially more natural peak limiting than most simple "limiters" can provide, resulting in improved performance when protecting broadcast STL's or power amplifiers in sound reinforcement systems.

Summary

The 424A "Studio Optimod" is the answer to many engineers' dreams. It combines a compressor, limiter, and de-esser in a most versatile way. Because its controls interact in a carefully human-engineered manner, it is easy and graceful to operate. Yet full flexibility is there to get the sound just right.

The professional audio and broadcast world has lived happily with "old favorite" limiter/compressors for a long time. If you examine the features, sound, performance, and price of our new "Studio Optimod" Model 422A/424A, we think you will agree that it is the new standard in dynamic range control. But we don't expect you to take our word for it. The proof is in the **listening.** We feel confident that once you A/B our unit against any of your current favorites, you will find a place for it in your rack. It's that good. A truly superior device, at the right price, at the right time.

Rest assured that with your new 422A/424A, you will continue to receive all the other things that you've come to expect from Orban products over the years — quality construction, comprehensive operating and service manuals, and unequalled customer service.

SPECIFICATIONS

INPUT
Impedance: greater than 10 k ohms active balanced; RF suppressed
LEVEL: -15dBm produces 10dB gain reduction with ATTACK TIME control centered, INPUT ATTEN control fully CW, and RATIO control at infinity-to-one

OUTPUT
Impedance: approximately 100 ohms, electronically balanced to ground; RF suppressed
Level: +4dBm nominal; absolute peak overload occurs at +26dBm

FREQUENCY RESPONSE
±0.25dB 20-20,000 Hz below limiting and de-esser thresholds

COMPRESSOR/LIMITER SECTION
Attack Time: manually adjustable in approximate range of 500us to 200ms; automatically scaled by program content
Release Time: adjustable in approximate range of 0.8dB/sec to 20dB/sec; automatically scaled by program content. Switch-selectable LINEAR and EXPONENTIAL release shapes.
Compression Ratio: adjustable from 2:1 to infinity-to-one at threshold. Lower ratios automatically increase beyond threshold.
Range Of Gain Reduction: 25dB
Tracking Of Multiple Channels: ±0.5dB
Total Harmonic Distortion (ATTACK and RELEASE TIME controls centered; infinite RATIO; 15dB gain reduction): less than 0.03% @1kHz. Typically 0.11% @20Hz; 0.02% @100Hz; 0.01% @1kHz; 0.04% @10kHz.
SMPTE IM Distortion (controls set as above; 60/7000Hz 4:1; 15dB gain reduction): typically 0.05%

DE-ESSER SECTION
Attack Time: approximately 1 ms
Release Time: approximately 30 ms
Harmonic Distortion: less than 0.05% THD introduced by de-essing action @10kHz
Available Gain Reduction: greater than 25dB

SYSTEM NOISE
RMS noise in 20-20kHz bandwidth better than 85dB below output clipping threshold for any degree of gain reduction; 90dB typical.

OPERATING CONTROLS
Compressor/Limiter
INPUT ATTENUATOR
COMPRESSION RATIO
ATTACK TIME
RELEASE TIME
RELEASE SHAPE
GATE THRESHOLD
OUTPUT TRIM
IDLE GAIN
COMPRESSOR/LIMITER OPERATE/DEFEAT
OUTPUT LEVEL (REAR PANEL)
De-Esser
THRESHOLD
DE-ESSER OPERATE/DEFEAT
General
STEREO COUPLING (424A only)
POWER ON/OFF

INDICATORS
Compressor/Limiter
GAIN REDUCTION METER
GATED LED
VCA LEVEL METER
De-Esser
NORMAL De-essing LED
HEAVY De-essing LED

Power Requirement
115/230 VAC ±10%; 50-60Hz. U-ground power cord attached; RF suppressed

Dimensions
19" (48.3cm) wide x 3.5" (8.9cm/2 units) high x 10" (25.4cm) deep

Operating Temperature
0-45 degrees C

Warranty
One year, parts and labor. Subject to limitations set forth in our Standard Warranty Agreement.

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The Orban 424A Gated Compressor/ Limiter/De-Esser

The Studio Optimod

Performance Highlights

- ☐ Intuitive and natural operation
- ☐ Adjustable attack time, release time, and compression ratio permit extremely natural processing or special effects
- ☐ Selectable linear (general purpose) or exponential (special purpose) release time characteristics
- ☐ Defeatable gate with adjustable threshold causes gain to move slowly toward user-adjustable value during pauses, preventing noise rush-up, pumping, or breathing
- ☐ Major controls interact to speed setup by keeping output levels relatively constant as controls are adjusted
- ☐ **Separate** compressor/limiter and de-esser control loops, each with optimized, program-controlled parameters
- ☐ Better than 25dB de-ess gain reduction available **in addition to** 25dB compressor/limiter gain reduction
- ☐ True peak-reading VCA LEVEL meter
- ☐ True peak-reading GAIN REDUCTION meter
- ☐ De-esser characteristics similar to highly-accepted Orban dedicated de-essers
- ☐ Low-distortion operation achieved using clean class-A VCA and distortion-cancelling control circuitry

Plus

- ☐ Rugged all-metal 19" rack mount package for ruggedness, roadworthiness, RFI shielding. Industrial-grade parts and construction
- ☐ Highly cost-effective; available in mono (model 422A) or stereo (model 424A)
- ☐ Multiple channels can be connected to track together
- ☐ Extensive RFI supression
- ☐ Balanced inputs and outputs, and 115/230V 50-60Hz power supply standard

THE 422A IS A SINGLE-CHANNEL VERSION OF THIS PRODUCT

The Model 422A/424A:

A multi-function Compressor/Limiter/De-Esser featuring exceptional versatility and ease of operation

Preface

We're about to say a lot about our 424A: we're proud of it. But no long essay can describe the bottom line — what it sounds like, and how it feels to use it. Comparisons of specifications and descriptions of new control techniques cannot describe the elusive and magical relationship between engineering and hearing.

This brochure should answer questions you might have about why the 424A is such a good-sounding and easy to use product. But when you get right down to it, the best way to appreciate its advanced design is to A/B it against your current favorite. Using it and listening to it are what really count!

The Orban 424A: It Had To Be Better, Or We Wouldn't Have Bothered

There are a lot of production limiters out there. Old favorites. Pretenders to the throne. The competition is fierce, and the market fragmented. So when Orban set out to design a significant new production limiter, we knew it had to be superior.

Fortunately, when we undertook the 424A R&D project, we had seven years of experience behind us. Enough to capture the Number One slot in the broadcast signal-processing market. Ask any FM broadcast engineer what the industry-standard limiter is, and he's likely to tell you it's our OPTIMOD-FM.

We've developed the technology — the **magic** which makes a compressor/limiter sound right, feel right, and operate quickly and intuitively. That's important in today's economic climate, where audio professionals demand fast, superlative results in recording studios, in broadcast production studios, and on the road doing demanding sound reinforcement work.

The result of our research and experience is the model 424A — a Gated Compressor/Limiter/De-esser with versatile features for production, and with a natural, transparent sound that has to be heard to be appreciated.

Our goal was to build a device which would produce the desired sound upon adjustment of a **minimum** number of controls. And, readjustment of one control should not require the corrective readjustment of others. We achieved this goal by making the ATTACK TIME, RELEASE TIME, and COMPRESSION RATIO controls interact with each other and with the threshold of limiting. For example, slower attack times permit more overshoots, so the threshold of limiting is automatically lowered to compensate. The result: you can concentrate on getting the **sound;** the mechanics largely take care of themselves.

The Un-Trendy Limiter

Most of the advertising buzzwords applied to AGC units are irrelevant to the essential listening qualities of these devices. Simultaneously, many of the truly important issues of AGC design (many of which are quite subtle and not easily reduced to buzzwords) seem to be unknown to, or ignored by, others.

The 424A is a very un-trendy device, in that it uses feedback control, an averaging detector, and a conventional "hard knee" static compression curve. Why?

For starters, because the type of detector (whether true-RMS, "linear-integration" averaging, or whatever) is essentially irrelevant, since far more variation results from simply changing the attack time than from changing the detector type. Similarly, the desirability of a consistent output as read on a VU meter is questionable, since VU meter readings have only the most casual correlation to psychoacoustical loudness.

Feedback control circuitry has been accused of hiding the vices of inferior VCA's while introducing instability and high amounts of distortion. Our VCA has nothing to hide, our loop is totally stable, and our measured THD is significantly better than most others on the market — feedback or feedforward. We use feedback because our ears tell us that, properly implemented, it creates a control loop dynamic response which simply **sounds better.**

The "soft knee" compression ratio characteristic promoted by others is a way of making a compressor sound innocuous by "sneaking up" upon high ratios over the course of many dB. Low ratios always sound more graceful than high ratios. In a "soft-knee" compressor, mostly low ratios are used. No wonder the sound of the compressor is improved! Alas, the price exacted is lost loudness and inconsistent level control. This may be fine for certain applications, but not if you're trying to persuade a wide dynamic range vocal to cut through a heavy instrumental backing, or if you're trying to make anything audible at all times.

What then, **does** make a good-sounding compressor/limiter? Primarily, the dynamic response of the control loop. In the 424A, attack and release times are always "automatic" (i.e., program-controlled); the ATTACK TIME and RELEASE TIME controls merely scale the processes faster or slower without giving up the advantages of the program control.

If a slower attack time is chosen, an automatic gain-riding (AGC) function is achieved. Faster attack times introduce more peak limiting. So, depending on attack time, the 424A can serve as a compressor, limiter, or both simultaneously.

What Makes The 424A So Special?

The Orban 424A has a real compression ratio control for those few times when you want to maintain some **real** dynamic range. However, you will probably be astonished at the "openness" and **apparent** dynamic range available even at the "infinite" ratio. You may be even more astonished when you discover the apparent loudness increase achieved without the usual side effects. Virtually any competent compressor can sound natural if it is quiet enough (i.e., if it doesn't actually work very hard). **The real magic is sounding loud and natural simultaneously, as the 424A does.**

After you've lived with a 424A for a while, we suspect that the dust will build up on the ratio control as you realize that, finally, here is a compressor which doesn't have to be cranked back to a 2:1 or 4:1 ratio (whether manually, by choice, or involuntarily, by means of a "soft knee") to sound good! For those of you who won't give up the "soft knee" no matter what, we've hedged our bets: you'll be pleased to know that at the lower settings of the 424A RATIO control, the ratio increases as more gain reduction is used.

There is also control over the **shape** of the release characteristic. Ordinarily, the release characteristic is "linear." (Recovery, in the absence of program material, proceeds at a constant number of dB per second.)

A switch-selectable "exponential" release shape is also provided for special applications. This forces the release to start slowly and to increase in speed as it proceeds.

While it sounds substantially less natural than "linear" on most program material, it can be useful when gain-riding wide-dynamic range single tracks (like vocalists) where the "open" sound of a slow release time is desired, yet quick gain-riding is necessary to make levels more consistent.

The IDLE GAIN control is a unique and unusually useful new feature. This control sets the gain of the compressor when it is in the "idle" mode. (When the compressor is manually defeated by operation of the DEFEAT switch, or, when it is gated by low-level audio or silence.)

In either case, the gain will move smoothly toward a value specified by the setting of the IDLE GAIN control — quickly after manual compressor defeat, and more slowly under gated conditions.

If the IDLE GAIN control is set close to the average gain reduction, then, upon resumption of ordinary compression, there will be minimum gain change in the VCA, and therefore minimum audible side effects will occur. The feature is extremely useful in preventing noise or tape hiss from being pumped up, and in facilitating smooth manual switching of the COMPRESSOR OPERATE/DEFEAT switch during program. Its effect is usually substantially smoother than that of a conventional noise gate.

The OUTPUT TRIM control can be used to force some peak clipping in the VCA in applications requiring tight control of peak levels (like the protection of a broadcast STL), thus controlling fast peaks without need for extremely fast attack times (which almost invariably cause more audible degradation than a modest amount of overshoot clipping). Conveniently, the peak-reading VCA LEVEL meter not only allows you to adjust the 424A to **clip if you want it to,** but also makes it easy to **avoid clipping entirely** if that is your goal.

About Distortion And VCA's

In any compressor/limiter, the static nonlinearities of the VCA are ordinarily totally overshadowed by the dynamic distortions caused by literally intermodulating the input signal by a rapidly varying control voltage. Sometimes such distortion is heard as additional unwanted spectral components, such as traditional harmonic and IM distortion. At other times, it is perceived as unnatural modulations of the signal peculiar to AGC devices such as "pumping," "hole-punching," "shivering," and a whole bunch more which no one has bothered to name, but which most musically-sensitive people can easily hear!

In the Orban 424A, nonlinear control voltage smoothing results in singularly favorable **dynamic** distortion properties. Our proprietary VCA complements this low dynamic distortion by slewing quickly, having "soft" low-order static distortion which is well below the threshold of audibility (due, in part, to class-A operation), and having noise performance which does not deteriorate as the amount of gain reduction increases.

The Final Coup: A Full-Function De-Esser

Finally, our de-esser. The VCA used in the 424A has **two** gain-control inputs. This means that we can add a **no-compromise** de-ess function which is essentially **independent** of the compressor/limiter. As we have proven in our dedicated de-essers, the optimum attack times and release times are quite different for the compressor/limiter and de-esser functions. In the 424A, each function is independently optimized.

The de-esser section of the 424A sounds about the same as our popular dedicated de-essers. It controls sibilance levels by quick broadband gain reduction when excessive "ess" energy is detected. In this way, any low-frequency IM distortion is reduced along with the "ess," and coloration of the "ess" sounds does not occur. One extra benefit of the de-esser section of the 424A is that it can effectively de-ess sibilant vocals which have already been **mixed** with other program, and, in this application, effectively acts more like a high-frequency limiter.

The only essential difference between the 424A de-esser and the Orban dedicated de-essers is that the 424A lacks the dedicated de-essers' ability to provide constant de-essing regardless of input level. In the 424A, this does not create a problem since it was assumed that the input to the de-esser section would be compressed by the 424A's compressor/limiter and would therefore be at a constant level. LED's indicating NORMAL/HEAVY de-essing allow the SENSITIVITY control of the de-esser to be readily adjusted.

Mono, Stereo, Or Dual-Channel

A STEREO COUPLING switch allows you to operate the two channels of the 424A independently or in stereo. In stereo, the channels will typically track within ±0.5dB. Rear-panel coupling terminals allow tracking an unlimited number of channels together.

Technical Description

The 672A Equalizer consists of a balanced input buffer amplifier, eight main equalization amplifiers connected in series, and tunable lowpass and highpass positive feedback 12dB/octave Butterworth filters. The output of the lowpass filter is buffered to drive 600 ohms, and is available separately. By suitable switch settings, the main output can be made to carry a highpassed signal. Thus the 672A can be used as an equalizer cascaded with a full electronic crossover, or as an equalizer cascaded with lowpass and highpass filters.

Each amplifier in the equalizer section provides equalization for one band only, thus assuring no interaction between bands. The total equalization is simply the sum (in dB) of the equalizations provided by each of the sections.

Peak boost is accomplished by adding the output of a two-pole bandpass resonator to the main signal; reciprocal dip occurs when this resonator is symmetrically connected as a feedback element in the main equalizer amplifier.

The EQ IN/OUT switch bypasses the last seven main amplifiers and defeats equalization in the first amplifier. Gain and signal polarity are equal in the IN and OUT modes. As the BANDWIDTH control is operated, the skirts of the equalization curve move in and out, but the peak gain and peak frequency remain constant. As the EQUALIZATION controls are operated, the frequencies of peak gain remain constant. However, as the TUNING control or EQUALIZATION control (in *dip* mode) are operated, the bandwidth ("Q") will change, because of the simplifications in the "quasi-parametric" bandpass resonator. Careful design has enabled us to produce curves (in boost mode only) essentially identical to the desirable "constant-Q" curves provided by our 622B *true parametric* equalizer in its *boost* mode.

The EQUALIZATION controls all produce peaking curves; if shelving curves are desired, they can be approximated by tuning the lowest band to 20Hz and the highest band to 20kHz. The breakpoint of the shelving characteristic is then adjusted with the BANDWIDTH control.

Summary

Many people are now aware of the *power* of parametric equalization: the almost sensual satisfaction of getting the sound really *right*. These same people are also demanding professionals, insisting on inaudible noise and distortion, human engineering, quality "feel", and uncompromising reliability.

Orban is well-known for its line of fine parametric equalizers, like the 622B. Now with the 672A, it brings equalization of the same rigorous quality to applications where it could never before be afforded. The 672A is inexpensive enough to qualify it for serious consideration in applications which would otherwise be given by default to a much less able graphic equalizer.

The 672A rounds out the line of Orban "Professionals' Parametrics." Between the 622B *true parametric* and the 672A *quasi-parametric*, there is an equalizer for virtually every need and budget. The Orban "Professionals' Parametrics" are available at your authorized Orban dealer.

Specifications: 672A Equalizer

All specifications apply when driving 600 ohms or higher impedances. Noise measured on an average-reading meter through a 20-20,000Hz bandpass filter with 18dB/octave Butterworth skirts.

Electrical

Input:
Impedance, Load (each leg): 100K in parallel with 1000pF, electronically balanced
Impedance, Driving: Ideally 600 ohms or less, balanced or unbalanced
Nominal Input Level: Between -10 and +4dBm
Absolute Overload Point: +26dBm

Output:
Impedance, Source: 47 ohms in parallel with 1000pF, unbalanced (Optional transformer balanced 600 ohm outputs)
Impedance, Load: Should be 600 ohms or greater-will not ring into any capacitive load
Nominal Output Level: +4dBm
Max. Output Level Before Clipping: +19dBm, 20-20,000Hz

Frequency Response:
±0.25dB, 20-20,000Hz: EQ controls set at "O" detents

Available Gain:
+12dB; adjustable to — infinity by means of front-panel GAIN control

Slew Rate:
Varies between 6 and 13V/uS depending upon setting of GAIN control; slewing is symmetrical. Internal bandlimiting assures that slew rate limiting will not occur even with the most severe equalization and program material.

Square Wave Response:
Square wave exhibits no spurious ringing at any output level. The only ringing observable is that theoretically associated with any given equalization curve.

Total Harmonic Distortion:
Less than 0.05%, 20-20,000Hz (+18dBm)

SMPTE Intermodulation Distortion:
Less than 0.05% (+18dBm: 60/7000Hz, 4:1)

Noise at Output:
Less than -84dBm (EQ In, filters out, controls centered)

Overload/Noise Ratio:
Better than 113dB for any single bandpass filter, for any settings of TUNING or BANDWIDTH controls.

Equalization Ranges:
±16dB peaking EQ, Reciprocal

Tuning Ranges:
20-60Hz; 40-150Hz; 110-310Hz; 230-750Hz; 480-1900Hz; 1.1-4.5Hz; 2.8-9.0kHz; 5.9-21kHz. Dials calibrated at ISO preferred frequencies.

"Q" Range:
Greater than 0.5 to 10 for any setting of the TUNING control

Low Pass Filter Section:
Tunable in 2 ranges: 200-2000Hz or 2.0-20kHz, 12dB/octave, (2nd-order Butterworth)

High Pass Filter Section:
Tunable in 2 ranges: 20-200Hz or 200-2000Hz, 12dB/octave, (2nd-order Butterworth)

Overload Circuit:
Lamp lights for 200mS if the instantaneous peak output of any amplifier rises to within 1dB of its clipping point.

Circuit Design:
Active RC realized with FET-input opamps. Line driver employs discrete transistor current booster.

Operating Temperature:
0-50°C

Power Requirements:
115/230VAC ±10%; 50/60Hz; 6 watts

Physical

Operating Controls:
EQUALIZATION, TUNING, and BANDWIDTH for each of eight bands. TUNING, RANGE (x1; x10), and FILTER IN/OUT for each filter. EQUALIZATION IN/OUT, POWER ON/OFF, and GAIN for entire equalizer.

Panel:
19" x 5¼" (48.3 x 13.3cm): 3 units

Chassis Depth Behind Panel:
5¼" (13.3cm)

Weight:
Net: 8 lbs. (3.6kg); Shipping: 12 lbs. (5.4kg)

AC Cord:
3-wire U-ground to USA Standard

Connectors:
140 type barrier strip (#5 screw) plus parallel-wired ¼" 3 ckt. phone jacks (Switchcraft 12B or equal). Holes punched for XLR-type connectors (Switchcraft D3F and D3M or equal)

Circuit Ground:
Available on barrier strip; normally jumpered to chassis.

Specifications subject to change without notice.

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The Orban 672A Equalizer:

more versatility than a graphic;
more economy than a full parametric.

Performance Highlights:

Graphic EQ Section

- Eight bands, each with TUNING and BANDWIDTH control
- Each band tunes over 3:1 frequency range
- "Q" typically variable between 0.3 and 20 (center TUNING)
- ± 16dB equalization range
- EQ controls are long-throw dust-shielded slidepots for good resolution
- TUNING and BANDWIDTH controls marked with "tics" indicating typical settings
- Narrowband notching capability ideal for sound reinforcement
- Bands totally non-interacting

HP/LP Filter Sections

- Each section continuously tunable over 100:1 range in 2 decades
- Each section independently switchable
- 12dB/octave slopes
- Filters follow graphic section. Separate main/highpass and lowpass outputs allow use as filters or as full electronic crossover

General

- Very low noise and distortion
- High slew rate for minimum TIM (SID)
- Front-panel GAIN control; 12dB gain available
- "Peak-stretching" overload lamp warns of clipping anywhere in equalizer
- Active balanced input; unbalanced outputs. Transformer-balanced outputs optional
- RF suppression on input, output, and power leads
- 115/230V, 50-60Hz transformer is standard
- Industrial-grade parts and construction including socketed IC's
- Highly cost-effective

Introducing the 672A Equalizer

The Orban 672A is a cost-effective, professional, quasi-parametric equalizer with the convenience of graphic-type EQ controls. Wide-range high- and low-pass filters with 12dB/octave Butterworth slopes follow the graphic section for added versatility. The 672A has two outputs, arranged so that these filters can also be used as a full tunable electronic crossover.

The 672A is a fully professional product designed to provide a large measure of versatility, convenience, and quality at a very attractive price. While it meets the requirements of the demanding professional, it is also designed and priced to make it understandable and available to the advanced audiophile.

To make the 672A easy to use in situations where its full versatility isn't needed, "tic" marks have been included on the dial calibrations of the TUNING and BANDWIDTH controls. When these controls are set to the tics, the 672A behaves like a standard octave-band graphic equalizer with the eight bands on ISO frequencies from 63 to 8000Hz.

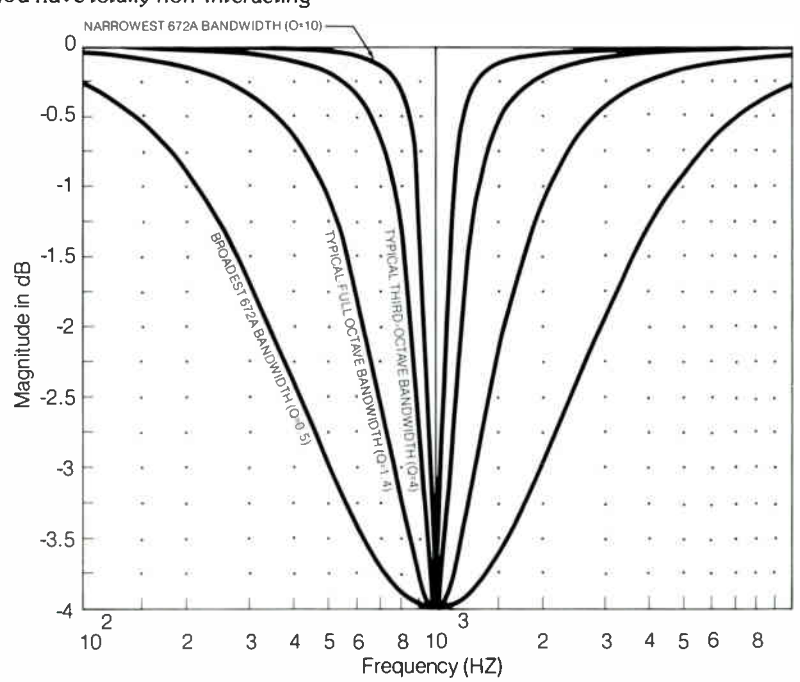
Each feature of the 672A has been thoughtfully chosen and cleverly implemented to make the equalizer a particularly powerful tool in nearly all areas of audio: sound reinforcement, public address, recording studio, broadcasting, motion picture sound, disco, theater . . .

Why "Quasi-Parametric"?

There are two basic types of parametric equalizer: *full-* and *quasi-* parametric. Orban manufactures both types. Both offer far more effective control than other kinds of equalizers, like graphics. Our popular dual-channel *622B* is a *full* parametric. This means that you have *totally non-interacting* control over the three fundamental *parameters* of equalization: the *amount* of peak boost or dip (in dB), the *tuning* (the frequency most affected by the equalization), and the *"Q"* (which relates to the sharpness of the EQ curve — the degree to which frequencies on either side of the peak frequency are affected by the equalization). As opposed to our *622B* Equalizer, the 672A is *quasi*-parametric. This means that the "Q" changes when you adjust the TUNING and/or EQ controls. Other control adjustments are completely non-interacting: TUNING and EQ do not affect each other.

The other important performance difference between the full-parametric 622B and the new 672A is that the 622B's EQ curves are *"constant-Q"*; the 672A's curves are "reciprocal." "Constant-Q" curves are valuable in that they permit infinite-depth notches to be created; reciprocal curves limit the maximum cut to the same number of dB as the maximum boost. In the case of the 672A, 16dB of cut is available. This is fine for tuning out ring-modes in sound reinforcement systems, but might not be adequate in all circumstances to remove hum or other fixed-frequency interference from a signal. On the other hand, some people prefer reciprocal curves because the boost and cut are mirror images of each other, thus permitting previous equalization to be readily "undone" later. Careful design of the circuitry gives the 672A *in boost mode* a characteristic similar to the 622B's desirable "constant-Q" curve family.

Why did we choose the quasi-parametric technique for the 672A? Because it offers a way to produce a very high quality, stable equalizer at low cost without compromising distortion, noise, accuracy, or reliability.



Various Bandwidth Dipping Curves — 4dB Dip

Applications:

Sound Reinforcement

There are many ways to use the 672A in sound reinforcement:

1) In an economy biamped installation, replace both the third-octave equalizer *and* the electronic crossover with the 672A. The 672A's narrowband, tunable notches can deal with ring modes more effectively than the third-octave unit could. Use three or four of the 672A bands for narrowband notching; leave the rest for wideband EQ.
2) In a higher budget biamped installation, use the 672A as an electronic crossover plus a narrowband, tunable notch filter for ring mode suppression; incorporate a separate third-octave equalizer to correct the house curve.
3) Use variations of (1) and (2) above *with* an electronic crossover; the 672A's high-pass and lowpass filters can then be used to roll off the frequency response of the system in a controlled manner.
4) In a *non-biamped* system (like a stage monitor), use the 672A to equalize the monitor, and use its filters to restrict response in the extreme high and low frequencies.
5) Use the 672A as a *partial* electronic crossover plus equalizer/filter by devoting one channel of 672A equalization to each driver; one filter is required to perform the crossover function; the other can be used for its normal highpass (or lowpass) function.

In all cases, the BANDWIDTH control can be adjusted to make the totally non-interacting (series-connected) bands "combine" — a most desirable characteristic in sound reinforcement.

Any way you cut it, the 672A's economy and extraordinary versatility make it one of the sound reinforcement practioner's most useful tools.

Recording Studios

Every recording studio needs a few channels of 672A equalization to handle the tough chores that the internal console equalizers can't deal with. Patch that problem track through a 672A: its fine-tuning ability lets you clean up the track far more effectively than you could with a graphic or "three knob" console equalizer. Use the tunable filters to help eliminate rumble, cymbal splash, kick drum leakage — you name it!

The 672A is also an ideal adjunct to an electronic music synthesizer — you can create high "Q" formants and shape the spectrum so that the sound comes alive.

If you need to correct the equalization of a finished track because of second thoughts after the mix, the 672A can create the finishing touches as no ordinary equalizer can. It's better than a third-octave graphic, because the 672A can generate broad, non-ringing boosts, whereas the graphic is much more colored and ringy.

Motion Picture Sound

The 672A is an ideal replacement for the graphic equalizers ordinarily used for dialogue equalization in motion picture sound. Set the TUNING and BANDWIDTH controls to the "tics" on the panel, and you get the equivalent of a familiar, easy-to-operate graphic. But when you *need* the extra control and flexibility — such as notching out the extraneous sounds that always seem to plague location recordings — that power is there instantly, without patching or the use of external dip filters. The high and lowpass filters are invaluable for cleaning up noise and rumble without affecting dialogue — and without using up EQ channels to try to achieve filtering. In addition, many "effects" (such as telephone, pocket radio, or "old time" recordings) can be easily created with the 672A alone.

In addition, the 672A can be used to equalize the "B-chain" in the re-recording theater to the acoustic response specifications of the studio. The lowpass filter can effectively simulate the "Academy Rolloff" or its current modifications.

Broadcasting

Use the 672A in the production studio to enhance the announce mike, and to create special production effects that make your station stand out among its competitors. Meanwhile, another 672A can be quietly and efficiently equalizing the program line for maximum punch and brightness on the air. Use the 672A to equalize phone or remote lines for flat response — it's much more versatile than the standard phone company equalizers. In the main studio, use it on the announce mike channel to equalize for maximum presence, and also to notch out sounds like mechanical hum from cart machine motors or air conditioning noise. Whatever your application, the 672A's RF suppression and optional output transformer mean problem-free installation in high-RF environments.

Discos

The 672A is an excellent disco equalizer. The sound contractor installing the system can offer the management exactly the sound desired — including solid, punchy bass free from muddiness and boom — and an aggressive, sizzling top free from the ringing and coloration typical of a full-octave graphic equalizer. The eight bands permit substantial work to be done in flattening out undesired response deviations in the upper bass and midrange. Narrowband notches can even deal with the difficult resonances sometimes encountered in high-efficiency horn-type loudspeakers. In biamped installations, use the separate lowpass and highpass filter outputs as a complete electronic crossover. No other crossover is necessary.

The 672A costs a bit more than an octave-type graphic. But, unlike a graphic, it really *solves* the problems.

(continued from inside)

5. "Hilbert-Transform Clipper": The Hilbert-Transform Clipper provides the peak limiting function and contains filters to assure that the clipping does not introduce out-of-band frequency components above 19KHz. Unlike a conventional audio clipper, its action introduces no harmonic distortion when it processes frequency components below 4kHz. Simultaneously, IM distortion below 2.2kHz is sharply cancelled by an adaptation of the patented Orban feedforward distortion-cancelling filter.

The result is very low perceived distortion on both voice and music. Voice is most severely degraded by harmonic, not IM, distortion. No harmonic distortion is produced in the voice frequency range, keeping voice clean. Sibilance distortion is eliminated by the distortion-cancelling filter. In the frequency range in which music has substantial energy (particularly after preemphasis), IM distortion is minimized, optimizing music reproduction as well.

Because bandlimited voice (from 16mm optical film, for example) is so prevalent in TV audio and because bandlimited voice is exceedingly sensitive to the harmonic distortion introduced by more conventional clippers, the Hilbert-Transform clipper is extremely effective in achieving cleaner sound in day-to-day operations.

6. Frequency-Contoured Sidechain (FCS) Overshoot Corrector: The Hilbert-Transform Clipper contains overshoots due to the addition of the distortion-cancelling signal and to unavoidable overshoots in its integral 15kHz lowpass filter. These overshoots are eliminated in the FCS Overshoot Corrector without adding out-of-band frequency components: The circuit acts essentially as a "bandlimited safety clipper".

Because this circuit acts instantaneously and employs no gain reduction or dynamic filtering, it causes neither pumping nor dulling of program material.

7. Noise Reduction Port, Output Amplifier, And Deemphasis: At the output of the Overshoot Corrector, the signal is peak-controlled and pre-emphasized. The L and R outputs of the Overshoot Corrector are applied to a matrix which produces L + R and L – R. Jumpers determine whether the OPTIMOD-TV Noise Reduction Port is fed L/R or L + R/L - R signals.

The Noise Reduction encoder can be bypassed by the Noise Reduction IN/OUT switch on the rear panel of OPTIMOD-TV. The output of the switch (which selects either the input line to the encoder or the output line from the encoder) is applied to a balanced transformerless line amplifier with strappable deemphasis.

Best system peak control is obtained by defeating exciter preemphasis and applying the preemphasized signal from OPTIMOD-TV to the flat exciter. In some exciters it is inconvenient to defeat the preemphasis, so the exciter must be supplied with a "flat" (i.e., deemphasized) signal from OPTIMOD-TV, which is readily accomplished by moving jumpers.

The outputs of the line amplifiers are interfaced to the outside world through non-overshooting RFI filters effective from approximately 500kHz to 1GHz.

Summary:

OPTIMOD-TV is an integrated "system approach" to

- ☐ ride gain
- ☐ perform compression as desired
- ☐ control excessive loudness
- ☐ control peaks by high-frequency limiting, distortion-cancelling "Hilbert-transform clipping", and bandlimited overshoot correction.

This optimizes technical parameters to their practical limit while producing a sound at the viewer's ear which is perceived as natural, pleasant, and free from the processing artifacts that often plague other signal processing approaches.

Order Guide:

8182A	OPTIMOD-TV AUDIO PROCESSING SYSTEM
OPT-18	50us preemphasis installed
8182A/ST	Studio Accessory Chassis for "split" configuration
RET-25	Retrofit Kit to convert OPTIMOD-TV Model 8180A to Model 8182A. Requires return to the factory for modification and alignment.
MAN-8	Additional Copy Of 8182A Operating Manual

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Specifications

FREQUENCY RESPONSE (System in PROOF mode)

Follows standard 75us preemphasis curve ±0.75dB, 50-15,000 Hz. (50us preemphasis available on special order.) Deemphasis jumper on line amplifier card permits flat output ±0.75dB 50-15,000Hz for use with external preemphasis. All preemphasis networks include a fourth-order lowpass filter and fourth-order phase corrector prior to the high-frequency limiter and clipper to prevent these elements from processing out-of-band program material and to minimize overshoot, thus minimizing the amount of high-frequency limiting and clipping.

INPUT CONDITIONING

Highpass Filter: Third-order Chebychev with 30Hz cutoff and 0.5dB passband ripple. Down 0.5dB at 30Hz; 10.5dB at 20Hz; 31.5dB at 10Hz. Protects against infrasonic destabilization of certain exciters' AFC's, as well as infrasonic gain modulation in the compressor.

Phase Scrambler: Allpass network makes peaks more symmetrical to best utilize the symmetrical peak overload characteristics of the FM medium.

NOISE

–75dB below 100% modulation, 50-15,000 Hz maximum; –81dB typical.

Total System Distortion (PROOF Mode; deemphasized; 100% Modulation)
Less than 0.25% THD, 50-15,000Hz (0.02% typical); less than 0.1% SMPTE Intermodulation Distortion (60/7000Hz; 4:1).
["THD" is defined as the root-sum-square (R.S.S.) sum of all harmonics, 50-30,000Hz. Noise (which is unavoidably included in the reading on a typical THD analyzer) is specifically excluded from this specification.]

"MASTER BAND COMPRESSOR CHARACTERISTICS

Attack Time: approximately 1ms.

Release Time: program-controlled—varies according to program dynamics and amount of gain reduction (see text). Process can be scaled fast or slow by means of continuously variable RELEASE TIME control. Employs delayed release for distortion reduction.

Total Harmonic Distortion (measured at VCA output, OPERATE Mode, RELEASE TIME control centered): Less than 0.1%, 200-15,000Hz, +10 to –15dB gain reduction.

Available Gain Reduction: 25dB

Metering: Three dB-linear edgewise-reading gain reduction meters—
TOTAL is true peak-reading with electronic acceleration and peak-hold (+10 to –15dB);
LIMITING indicates fast peak limiting component of gain reduction (0-5dB);
COMPRESSION indicates slow compression component of gain reduction (+10 to –15dB).

Gain Control Element: True VCA. Proprietary Class-A design eliminates crossover notch distortion, modulation noise, and slewrate limiting found in competitive Class-AB designs.

"BASS" BAND COMPRESSOR CHARACTERISTICS

Attack Time: program-controlled; not adjustable.

Release Time: program-controlled; not adjustable. Incorporates delayed-release distortion reduction.

Total Harmonic Distortion (at VCA output, OPERATE mode):
Less than 0.1% THD, 50-200Hz, +10 to –20dB gain reduction.

Available Gain Reduction: 30dB

Metering: single dB-linear edgewise-reading gain reduction meter (+10 to –20dB).

Gain Reduction Element: Proprietary Class-A true VCA.

Bass Coupling (U.S. patent #4,249,042): Enables gain of "Bass' band to track gain of "Master" band to any degree, from identical tracking to fully independent operation. Adjustable with BASS COUPLING control.

CROSSOVER CHARACTERISTICS

Control: 6dB/octave @200Hz;

Program: 12dB/octave @200Hz in unique "distributed crossover" configuration (U.S. patent #4,249,042).

HIGH FREQUENCY LIMITER CHARACTERISTICS

Attack Time: approximately 5ms.

Release Time: approximately 20ms. Delayed release included for distortion reduction.

Mode: Left and right channels operate independently to avoid high frequencies in one channel causing audible timbre modulation of opposite channel.

Control Elements: Junction FET

Metering: Two LED's indicate HF limiting in L and R channels.

Threshold of HF Limiting: User-adjustable over 3dB range to meet format requirements.

HILBERT-TRANSFORM CLIPPER CHARACTERISTICS

Nominal Bandwidth: 15.4kHz

Distortion Characteristics: Less than 2.5% THD is produced by individual frequencies 30-4000Hz when driving the Hilbert-Transform Clipper to 6dB beyond its threshold of limiting. With drive frequencies above 4kHz, the characteristics revert to those of a very "hard" conventional clipper. Further distortion cancellation assures that, for any arbitrary input (including program material), distortion components in the frequency range from 0-2.2kHz are cancelled better than 30dB below overshoot compensator threshold (patent pending).

Delay Correction: Fourth-order allpass.

Amount of Clipping: User-adjustable over 6dB range to match format requirements.

FREQUENCY-CONTOURED SIDE CHAIN (FCS) OVERSHOOT COMPENSATOR CHARACTERISTICS (patent pending)

System Overshoot: The FCS circuit is best thought of as a "bandlimited safety clipper". It operates like a hard clipper, but does not produce out-of-band frequency components as a simple hard clipper would. Because the audio processing will sometimes limit steady-state material with high average energy (like sinewaves) or with very little high-frequency energy to levels below the threshold of clipping, it is difficult to state a clear and meaningful specification for the *system* overshoot performance of the FCS circuit.
The FCS circuit is followed by a safety clipper. The overshoot specification could be slightly improved if this safety clipper were set up to clip more frequently. However, the system is aligned at the factory such that the safety clipper is almost *never* active, thus fully preserving the bandlimiting provided by the FCS circuit. With this safety clipper alignment, the peak modulation will be controlled ±3.5% on arbitrary waveforms clipped to any degree by the FCS circuit (acting as a bandlimited safety clipper); peak modulation will not exceed this level on other material. With typical program material, peak modulation uncertainty is less than 2%.

Sinewave Modulation Ability: 93% modulation (i.e., 0.6dB below maximum overshoot level) at all sinewave frequencies, assuming sinewaves are applied to FCS input.

Dynamic Separation: better than 45dB.

Difference-Frequency Intermodulation: FCS circuit causes no more audible IM (such as sibilance splatter) than would a simple hard clipper clipping to the same depth. The entire OPTIMOD-TV processing system is specifically configured to prevent the FCS circuit from audibly degrading the difference-frequency distortion-cancellation properties of the earlier peak limiting system.

SYSTEM SEPARATION

Greater than 50dB, 50-15,000Hz; 60dB typical.

INPUT

Impedance: greater than 10K ohms, electronically balanced by means of true instrumentation amplifier. Requires balanced source <= 600ohms.

Common Mode Rejection: Greater than 60dB @60Hz.

Sensitivity: –10dBm produces 10dB "Master" Band gain reduction @1kHz. Removal of internal 20dB pad permits –30dBm to produce same effect.

Connector: Barrier strip (#6 screw).

OUTPUT

Source Impedance: 370 ohms, independent of OUTPUT ATTEN setting, balanced.

Recommended Load Impedance: 600 ohms ±20%.

Level: variable –infinity dBm to greater than +20dBm by means of 15-turn OUTPUT ATTEN controls.

Connector: Barrier strip (#6 screw). RF suppressed.

TEST JACKS (for Test use only)

Provides L and R lowpass filter output on RCA phono-type connectors on rear panel. Outputs are unbalanced.

OPERATING CONTROLS

VU Meter Selector; switches ASA-standard VU meter to read:

L or R Input Level	(L INPUT BUFFER)
L or R Compressor Output	(L COMPR OUT)
L or R Filter Out	(L FILTER)
L or R Line Amplifier Output	(L SYSTEM OUT)
±15 V Power Supply Voltages	

SETUP CONTROLS (front-panel, behind lockable swing-down door—see Fig. 4-5)

Compressor:
- Left and Right Input Attenuators
- "Master" Band Release Time
- Release Shape Switch
- Gate Threshold
- Bass Coupling
- Clipping
- High-Frequency Limiter Threshold

General:
- Left and Right Output Attenuators
- PROOF/OPERATE Switches (to defeat gain reduction, HF limiting, clipping and gating)
- Loudness Controller ON/OFF Switch
- Power ON/OFF Switch
- 115V/230V Selector Switch
- Noise Reduction IN/OUT Switch (real panel)

POWER REQUIREMENT

115/230VAC, ±15%, 50-60Hz, approx. 31VA. IEC mains connector with detachable 3-wire "U-Ground" power cord supplied. Leakage to chassis less than 0.5mA. AC is RF-suppressed.

DIMENSIONS

19" (48.3cm)W × 7" (17.8cm) H × 12.5" (31.2cm)D—4 rack units

ENVIRONMENTAL

Operating Temperature Range: 0-50°C (32-122°F).

Humidity: 0-95% R.H., non-condensing.

WARRANTY

One year, parts and labor. Subject to limitations set forth in our Standard Warranty.

All specifications subject to change without notice.

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OPTIMOD-TV® 8182A

MONO OR STEREO

Function of OPTIMOD-TV:

Model 8182A is Orban's second-generation TV audio processor: It is an integrated signal-processing *system* which replaces conventional compressors, limiters, and clippers to precisely control audio modulation without introducing audible artifacts. Based on the popular OPTIMOD-FM Model 8100A, the 8182A offers the TV broadcaster the same superb quality that has made the 8100A so popular among FM radio broadcasters, plus Loudness Control and Hilbert-Transform clipping—features which tailor it perfectly to the unique requirements of television audio.

The 8182A is also ideally suited for conditioning signals prior to satellite uplinks, as well as for audio processing in other specialized communications applications where extremely high audio quality is desired, the audio spectrum must be limited to a 15kHz bandwidth, and the channel peak overload point is abrupt and must not be exceeded.

This brochure provides a technical description. However, it can't adequately describe the most important feature of the 8182A: its natural *sound*, and its ability to handle typical television audio feeds—from master tape to 16mm optical film—smoothly and gracefully without introducing processing artifacts. These days, consumers are accustomed to good sound, and OPTIMOD-TV helps you provide audio quality which augments the video quality you have worked so hard to achieve.

Briefly, the OPTIMOD-TV system performs these functions:

☐ It rides gain over a range of as much as 25dB, compressing dynamic range and compensating for gain-riding errors. Gain riding and compression are virtually undetectable because of advanced program-controlled time constants, level-dependent gating, and multi-band compression.

☐ It controls excessive perceived loudness by means of a complex loudness estimating circuit (which can be enabled and defeated by remote control). This circuit, licensed from CBS Technology Center, incorporates the results of their second major loudness research project (1978-1980). On-air tests of the controller have resulted in a substantial reduction or elimination of viewer complaints regarding excessively loud commercials.

☐ It controls potential interference to video and/or future stereo services by means of bandwidth-limiting 15kHz lowpass filters incorporating full overshoot compensation. OPTIMOD-TV thus provides extremely tight control over peak modulation, preventing overmodulation and controlling its output spectrum simultaneously.

☐ The OPTIMOD-TV compressor is a dual-band design which can be operated with the bands independent of each other ("independent"), or such that the bands are coupled and ordinarily track each other ("wideband"). When operated in "independent" mode, OPTIMOD-TV makes audio quality more consistent by correcting frequency balances between bass and midrange material. When operated in "wideband" mode, it will preserve frequency balances and will produce an output which sounds like its input.

☐ It prevents peak overload and overmodulation due to the effects of the preemphasis curve.

An accessory port is included to interface the noise reduction encoder required for TV stereo. In addition, an external TV stereo generator will be needed.

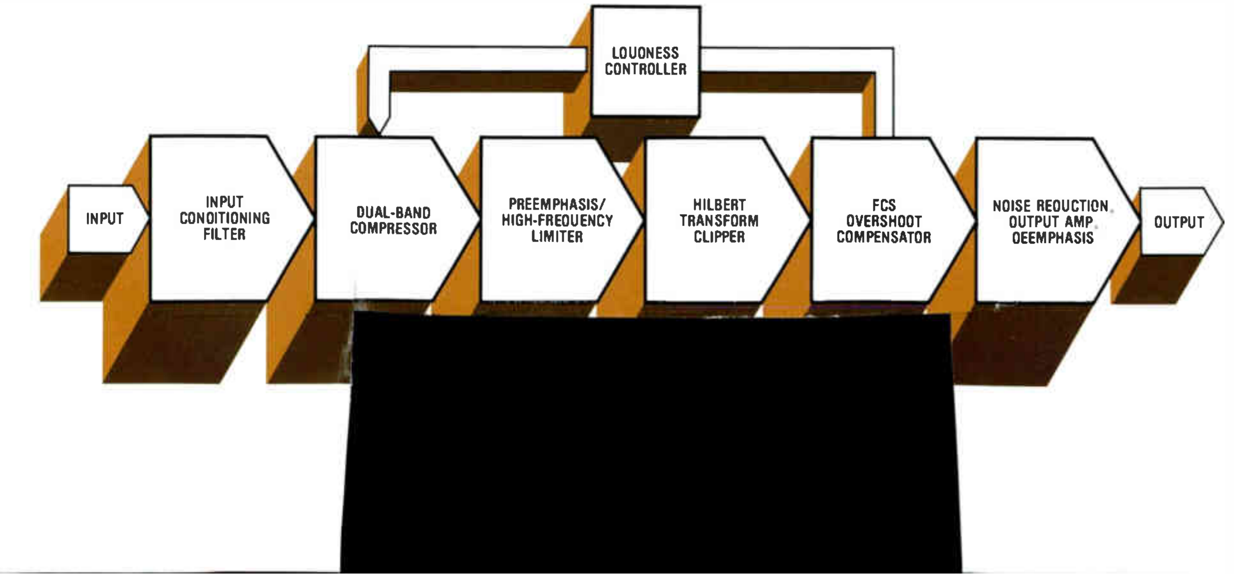
Internal jumpers determine if the active-balanced ±10dBm outputs are to be flat or preemphasized and whether they are to be in conventional left/right or in sum-and-difference form.

Ordinarily OPTIMOD-TV should be fed unprocessed audio. Additional compression and/or other audio processing would not be desirable except as might be applied to individual microphone channels in a live production environment or to other sources requiring special processing.

Split Configuration: An alternate, dual-chassis configuration permits the Dual-Band Compressor to be operated separately from the remainder of the circuitry. The Dual-Band Compressor can be placed at the studio side of the STL (telephone line, dual microwave, or FM subcarrier on a video STL) to protect the STL from overmodulation and to put most operating controls at the studio.

This configuration consists of a basic chassis in conjunction with the Model 8182A/ST Accessory Chassis. The 8182A/ST accepts several cards from the main chassis which are replaced by jumper cards.

However we discourage use of the split configuration if the gain of the STL cannot be maintained ±0.75dB. In this case we recommend an Orban Compressor/Limiter (such as the 424A) at the studio side of the STL to protect against STL overload.



Simplified System Description:

OPTIMOD-TV consists of seven basic blocks:

1. Input Conditioning Filter: An allpass phase scrambler to make peaks more symmetrical (thus reducing clipping distortion and permitting higher loudness), and a 30Hz 18dB/octave highpass filter to prevent subsonic information from disturbing the operation of the audio processing or exciters' AFC's.

2. Dual-Band Compressor: A "Bass" compressor which processes audio below 200Hz (12dB/octave crossover) in parallel with a "Master" compressor for the remainder. A BASS COUPLING control determines if the two bands operate discriminately ("independent" mode), or if the "Bass" band will be forced to track the "Master" band ("wideband" mode), preserving frequency balances. Intermediate bass coupling settings are also available.

Because of the unique design, the preemphasized output of the compressor can be directly applied to the peak limiting system: No further gain reduction is required for distortion control, and maximum naturalness is preserved.

Gating is provided to prevent noise rush-up during pauses (particularly with noisy 16mm optical sound tracks) and to make the 25dB gain reduction range more useful. The gating circuit is designed so that the gain does not get "stuck" forever in the 0 to 15dB gain reduction region—low-level program material is very slowly and imperceptibly increased in level even when gating is enabled.

The output level of the compressor is set by the CLIPPING control. This control thus sets the drive level to the subsequent high-frequency limiter and clipper, determining the amount of limiting and clipping.

3. Preemphasis And High Frequency Limiter: The output of the Dual-Band Compressor is applied to a phase corrector, 15kHz lowpass filter, preemphasis network, and high-frequency limiter.

The lowpass filter prevents out-of-band components from affecting the operation of the high-frequency limiter and avoids intermodulation between out-of-band frequency components and in-band frequency components in the clipper.

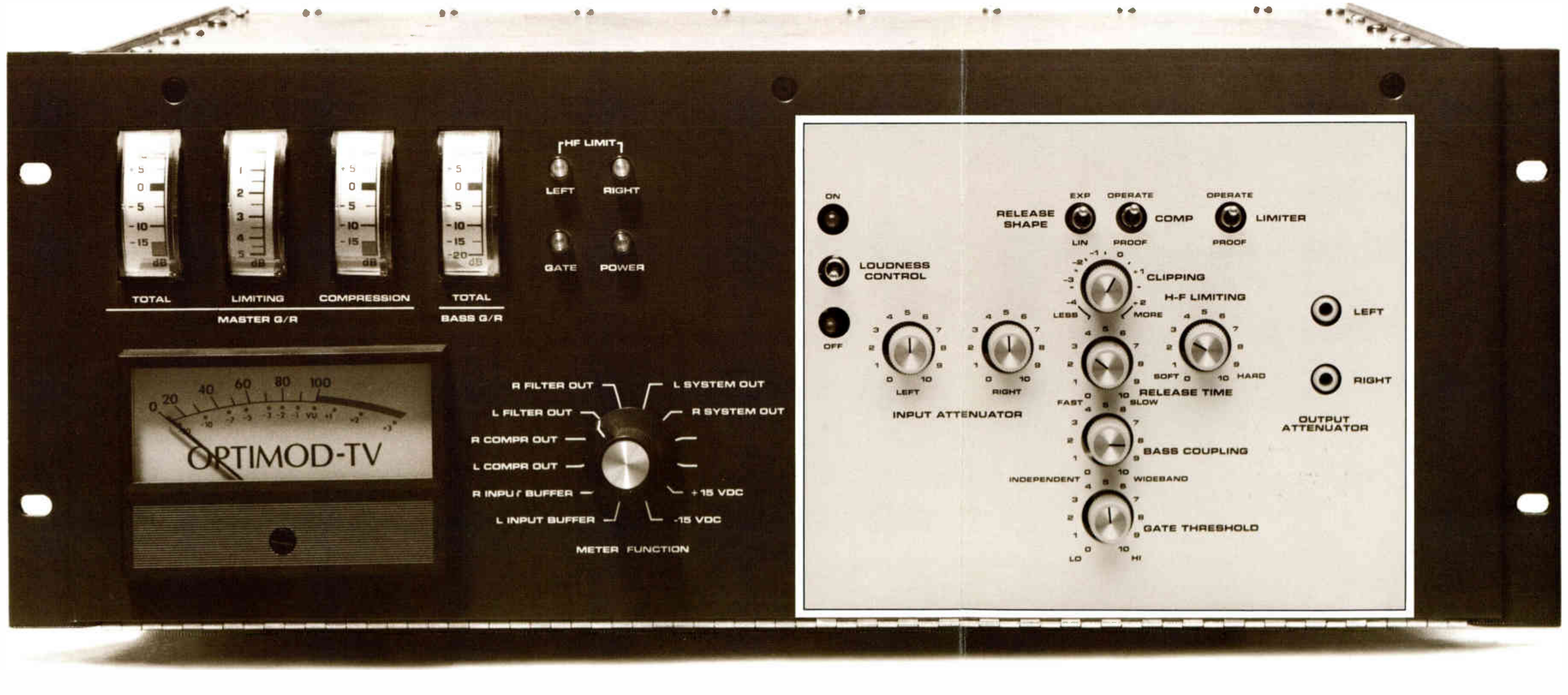
The high-frequency limiter is controlled by high frequencies *only* (rather than by the peak level of the preemphasized signal), eliminating any possibility of modulation of high frequency content by low frequency material. Its threshold of limiting is user-adjustable over a 3dB range, permitting brightness and high frequency distortion to be traded off according to your needs. Because the peak limiting system incorporates IM distortion cancellation, substantially more clipping can be accomplished without objectionable distortion than in conventional systems: Significantly improved high frequency power handling capability is achieved.

4. Loudness Controller: The concept of Loudness is different from the concept of Level. Loudness is *subjective* sound intensity: It is what the listener *perceives in his mind.* Level, on the other hand, can be measured in many objective ways: a VU meter and a PPM are two common level indicators in broadcast. No simple electrical measurement correlates well with loudness.

CBS Technology Center, in 20 years of experiments, has developed a technique of measuring loudness by means of a complex algorithm. This technique provides results which correlate well to loudness as subjectively judged by panels of listeners in extensive tests.

Ordinarily, gain reduction in OPTIMOD-TV is determined by the compressor's control circuitry. However, if the loudness exceeds a present threshold, the Loudness Controller acts to further reduce the gain as necessary. This is the most advanced technique known for measuring and controlling the loudness of broadcast audio.

Because certain sounds in entertainment programming (pistol shots, explosions, or screeching tires, for example) are *supposed* to be loud for dramatic impact, the Loudness Controller can be turned on and off locally or by remote control. ➤



SETUP CONTROLS
(BEHIND ACCESS DOOR)

672A Equalizer

A single channel quasi-parametric equalizer with continuous control over center frequency, bandwidth, and amount of peak or dip. Convenient graphic-style EQ controls provide reciprocal EQ in eight bands. Additional 12dB/octave highpass and lowpass filters tune continuously over 100:1 frequency range. Additional lowpass output permits use as equalizer cascaded with electronic crossover. GAIN control; overload lamp; IN/OUT switches for equalizer and each filter. Line level balanced input; unbalanced outputs can be balanced with optional transformer(s).

674A Equalizer

A two-channel version of the 672A. Controls are configured to facilitate accurate, easy adjustment of both channels simultaneously when equalizing stereophonic program. Each channel identical to the 672A, including all controls and overload indicator, with the exception that the electronic crossover outputs are arranged as "MAIN/LOWPASS" and "HIGHPASS" to provide further protection against accidental tweeter burnout should IN/OUT switches be incorrectly operated.

All products are equipped with 115/230V 50-60Hz power supplies and carry a One-Year Limited Warranty covering parts and labor. All products available through franchised dealers worldwide.

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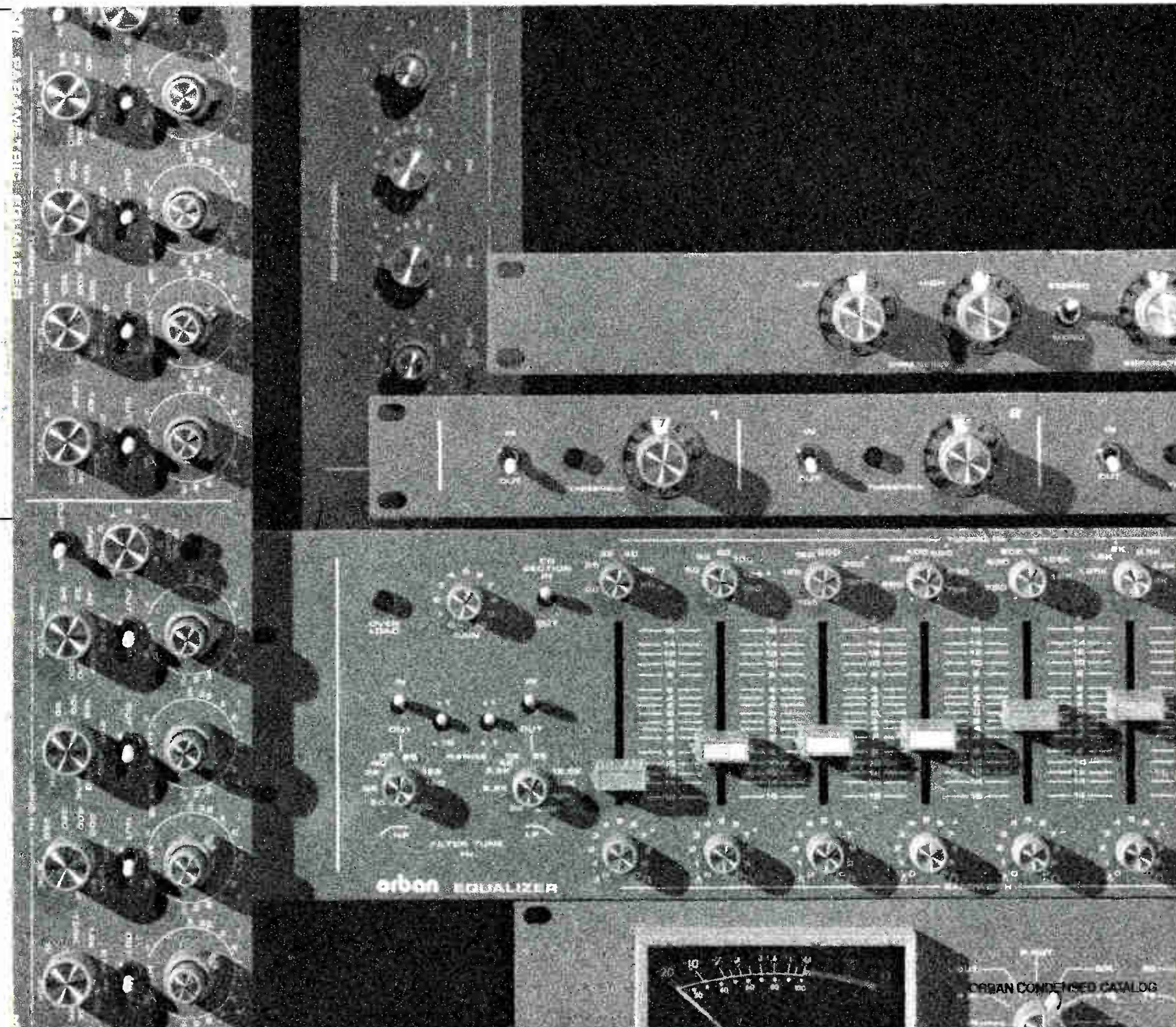
Orban Associates Inc., 645 Bryant Street, San Francisco, CA 94107 U.S.A.
Telephone: (415) 957-1067 Telex: 17-1480 Cable: ORBANAUDIO

Orban Condensed Catalog

Professional Audio Products

All Orban products are designed to meet the requirements of the most demanding professional users.

Detailed brochures which include full specifications are available on request from Orban dealers worldwide.



111B Reverberation

Dual-channel spring reverb with six springs/channel for smoothness and natural sound. "Floating threshold" limiter attenuates "spring twang" and protects against overload. Shelving bass and quasi-parametric midrange EQ. Unbalanced input accepts line-level or semi-pro (medium level) gear. Transformer-balanced main output; unbalanced "mixed" output allows use "in-line" without external mixers. Compact, rugged, and reliable.

516EC Dynamic Sibilance Controller

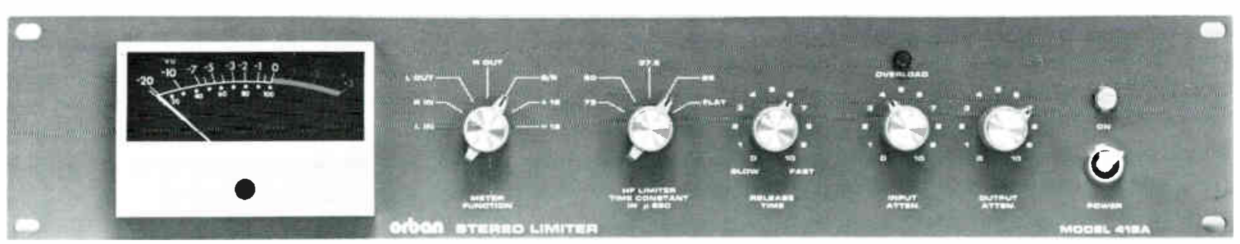
Three-channel de-esser with unbalanced line-level inputs and outputs. Ideal for simultaneous de-essing of several voices in cinema, recording, or broadcast. Easy to adjust with single THRESHOLD control. De-essing constant over 15dB input range. De-essing defeatable without clicks or pops. Outstandingly quiet and clean.

245E Stereo Synthesizer

Creates a seductive pseudo-stereo effect from mono originals. Left and right channels sum back to original mono for total compatability in disc cutting and FM stereo broadcast. Doesn't affect the frequency balance of the mono original. Easy to use; only three operating controls. Unbalanced line-level input and outputs.

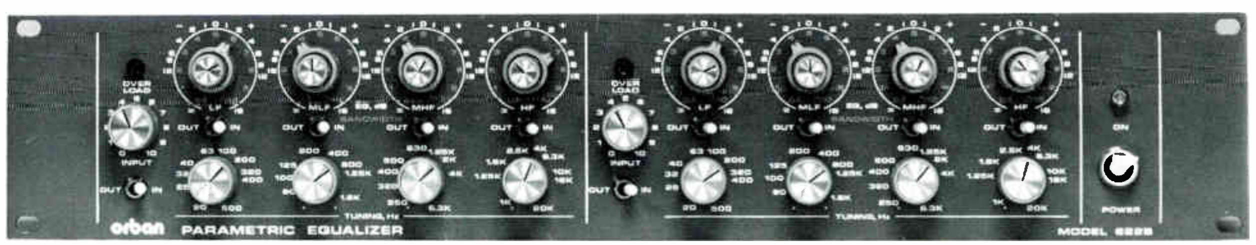
526A Dynamic Sibilance Controller

Effective de-essing without audible "action". De-essing constant over 15dB input level range. Easy to set up and use—only two operating controls (GAIN and THRESHOLD). Fully balanced transformer-coupled input and output with mic/line switching on input. Can be inserted and removed from circuit without clicks or pops. Convenient LED level and de-essing indicators.



418A Stereo Compressor/Limiter

Famous OPTIMOD-FM circuitry adapted for production and recording applications. Exceptionally smooth sound with adjustable program-controlled release time. Separate high frequency limiter with four selectable breakpoints from 25 to 75us. Simple, easy-to-use stereo-ganged controls. Accurate stereo tracking. Balanced line-level inputs; unbalanced outputs. Ideal for processing complex program material in cassette duplication, broadcast production, and recording studios.



622B Parametric Equalizer

A two channel parametric equalizer for use where continuous, non-interacting control over center frequency, bandwidth, and amount of peak boost or cut is desired. Four peaking bands per channel with "constant-Q" curves providing notching capability; individual channel and band in/out switches; GAIN control; overload lamp. Line-level balanced input and unbalanced output. Output can be balanced by addition of optional transformer.

Applications

The 424A can be applied anywhere that current compressors, limiters, and/or de-essers are used, since its versatility does not limit it to a single "sound," but instead lets it operate as a smooth, gentle compressor, a peak limiter, a de-esser, or any combination of these—all with a singular absence of undesirable artifacts.

This means that the unit can be used in recording studios, in broadcast production studios, ahead of broadcast studio-transmitter links, in sound reinforcement, and in video production and sweetening. It is an ideal all-in-one vocal processor, combining the necessary AGC and de-esser functions. It also shines when smoothly handling **mixed** program material—making it excellent for preparing cassette duplicating masters, or protecting tape recorders and cart machines from overload in tight time-pressure situations in broadcast and recording. Simultaneously, its versatile, wide-range setup controls make it a natural for processing single instrumental or vocal tracks in studios. Exploitation of the VCA clipping feature can result in substantially more natural peak limiting than most simple "limiters" can provide, resulting in improved performance when protecting broadcast STL's or power amplifiers in sound reinforcement systems.

Summary

The 424A "Studio Optimod" is the answer to many engineers' dreams. It combines a compressor, limiter, and de-esser in a most versatile way. Because its controls interact in a carefully human-engineered manner, it is easy and graceful to operate. Yet full flexibility is there to get the sound just right.

The professional audio and broadcast world has lived happily with "old favorite" limiter/compressors for a long time. If you examine the features, sound, performance, and price of our new "Studio Optimod" Model 422A/424A, we think you will agree that it is the new standard in dynamic range control. But we don't expect you to take our word for it. The proof is in the **listening.** We feel confident that once you A/B our unit against any of your current favorites, you will find a place for it in your rack. It's that good. A truly superior device, at the right price, at the right time.

Rest assured that with your new 422A/424A, you will continue to receive all the other things that you've come to expect from Orban products over the years—quality construction, comprehensive operating and service manuals, and unequalled customer service.

SPECIFICATIONS

INPUT
Impedance: greater than 10 k ohms active balanced; RF suppressed
LEVEL: −15dBm produces 10dB gain reduction with ATTACK TIME control centered, INPUT ATTEN control fully CW, and RATIO control at infinity-to-one

OUTPUT
Impedance: approximately 100 ohms, electronically balanced to ground; RF suppressed
Level: +4dBm nominal; absolute peak overload occurs at +26dBm

FREQUENCY RESPONSE
±0.25dB 20-20,000 Hz below limiting and de-esser thresholds

COMPRESSOR/LIMITER SECTION
Attack Time: manually adjustable in approximate range of 500us to 200ms; automatically scaled by program content
Release Time: adjustable in approximate range of 0.8dB/sec to 20dB/sec; automatically scaled by program content. Switch-selectable LINEAR and EXPONENTIAL release shapes.
Compression Ratio: adjustable from 2:1 to infinity-to-one at threshold. Lower ratios automatically increase beyond threshold.
Range Of Gain Reduction: 25dB
Tracking Of Multiple Channels: ±0.5dB
Total Harmonic Distortion (ATTACK and RELEASE TIME controls centered; infinite RATIO; 15dB gain reduction): less than 0.03% @1kHz. Typically 0.11% @20Hz; 0.02% @100Hz; 0.01% @1kHz; 0.04% @10kHz.
SMPTE IM Distortion (controls set as above; 60/7000Hz 4:1; 15dB gain reduction): typically 0.05%

DE-ESSER SECTION
Attack Time: approximately 1 ms
Release Time: approximately 30 ms
Harmonic Distortion: less than 0.05% THD introduced by de-essing action @10kHz
Available Gain Reduction: greater than 25dB

SYSTEM NOISE
RMS noise in 20-20kHz bandwidth better than 85dB below output clipping threshold for any degree of gain reduction; 90dB typical.

OPERATING CONTROLS
Compressor/Limiter
INPUT ATTENUATOR
COMPRESSION RATIO
ATTACK TIME
RELEASE TIME
RELEASE SHAPE
GATE THRESHOLD
OUTPUT TRIM
IDLE GAIN
COMPRESSOR/LIMITER OPERATE/DEFEAT
OUTPUT LEVEL (REAR PANEL)
De-Esser
THRESHOLD
DE-ESSER OPERATE/DEFEAT
General
STEREO COUPLING (424A only)
POWER ON/OFF

INDICATORS
Compressor/Limiter
GAIN REDUCTION METER
GATED LED
VCA LEVEL METER
De-Esser
NORMAL De-essing LED
HEAVY De-essing LED

Power Requirement
115/230 VAC ±10%; 50-60Hz. U-ground power cord attached; RF suppressed

Dimensions
19" (48.3cm) wide x 3.5" (8.9cm/2 units) high x 10" (25.4cm) deep

Operating Temperature
0-45 degrees C

Warranty
One year, parts and labor. Subject to limitations set forth in our Standard Warranty Agreement.

orban

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800017-000-02

The Orban 424A Gated Compressor/Limiter/De-Esser

The Studio Optimod

Performance Highlights

- ☐ Intuitive and natural operation
- ☐ Adjustable attack time, release time, and compression ratio permit extremely natural processing or special effects
- ☐ Selectable linear (general purpose) or exponential (special purpose) release time characteristics
- ☐ Defeatable gate with adjustable threshold causes gain to move slowly toward user-adjustable value during pauses, preventing noise rush-up, pumping, or breathing
- ☐ Major controls interact to speed setup by keeping output levels relatively constant as controls are adjusted
- ☐ **Separate** compressor/limiter and de-esser control loops, each with optimized, program-controlled parameters
- ☐ Better than 25dB de-ess gain reduction available **in addition to** 25dB compressor/limiter gain reduction
- ☐ True peak-reading VCA LEVEL meter
- ☐ True peak-reading GAIN REDUCTION meter
- ☐ De-esser characteristics similar to highly-accepted Orban dedicated de-essers
- ☐ Low-distortion operation achieved using clean class-A VCA and distortion-cancelling control circuitry

Plus

- ☐ Rugged all-metal 19" rack mount package for ruggedness, roadworthiness, RFI shielding. Industrial-grade parts and construction
- ☐ Highly cost-effective; available in mono (model 422A) or stereo (model 424A)
- ☐ Multiple channels can be connected to track together
- ☐ Extensive RFI supression
- ☐ Balanced inputs and outputs, and 115/230V 50-60Hz power supply standard

THE 422A IS A SINGLE-CHANNEL VERSION OF THIS PRODUCT

The Model 422A/424A:

A multi-function Compressor/Limiter/De-Esser featuring exceptional versatility and ease of operation

Preface

We're about to say a lot about our 424A: we're proud of it. But no long essay can describe the bottom line—what it sounds like, and how it feels to use it. Comparisons of specifications and descriptions of new control techniques cannot describe the elusive and magical relationship between engineering and hearing.

This brochure should answer questions you might have about why the 424A is such a good-sounding and easy to use product. But when you get right down to it, the best way to appreciate its advanced design is to A/B it against your current favorite. Using it and listening to it are what really count!

The Orban 424A: It Had To Be Better, Or We Wouldn't Have Bothered

There are a lot of production limiters out there. Old favorites. Pretenders to the throne. The competition is fierce, and the market fragmented. So when Orban set out to design a significant new production limiter, we knew it had to be superior.

Fortunately, when we undertook the 424A R&D project, we had seven years of experience behind us. Enough to capture the Number One slot in the broadcast signal-processing market. Ask any FM broadcast engineer what the industry-standard limiter is, and he's likely to tell you it's our OPTIMOD-FM.

We've developed the technology—the **magic** which makes a compressor/limiter sound right, feel right, and operate quickly and intuitively. That's important in today's economic climate, where audio professionals demand fast, superlative results in recording studios, in broadcast production studios, and on the road doing demanding sound reinforcement work.

The result of our research and experience is the model 424A—a Gated Compressor/Limiter/De-esser with versatile features for production, and with a natural, transparent sound that has to be heard to be appreciated.

Our goal was to build a device which would produce the desired sound upon adjustment of a **minimum** number of controls. And, readjustment of one control should not require the corrective readjustment of others. We achieved this goal by making the ATTACK TIME, RELEASE TIME, and COMPRESSION RATIO controls interact with each other and with the threshold of limiting. For example, slower attack times permit more overshoots, so the threshold of limiting is automatically lowered to compensate. The result: you can concentrate on getting the **sound;** the mechanics largely take care of themselves.

The Un-Trendy Limiter

Most of the advertising buzzwords applied to AGC units are irrelevant to the essential listening qualities of these devices. Simultaneously, many of the truly important issues of AGC design (many of which are quite subtle and not easily reduced to buzzwords) seem to be unknown to, or ignored by, others.

The 424A is a very un-trendy device, in that it uses feedback control, an averaging detector, and a conventional "hard knee" static compression curve. Why?

For starters, because the type of detector (whether true-RMS, "linear-integration" averaging, or whatever) is essentially irrelevant, since far more variation results from simply changing the attack time than from changing the detector type. Similarly, the desirability of a consistent output as read on a VU meter is questionable, since VU meter readings have only the most casual correlation to psychoacoustical loudness.

Feedback control circuitry has been accused of hiding the vices of inferior VCA's while introducing instability and high amounts of distortion. Our VCA has nothing to hide, our loop is totally stable, and our measured THD is significantly better than most others on the market—feedback or feedforward. We use feedback because our ears tell us that, properly implemented, it creates a control loop dynamic response which simply **sounds better.**

The "soft knee" compression ratio characteristic promoted by others is a way of making a compressor sound innocuous by "sneaking up" upon high ratios over the course of many dB. Low ratios always sound more graceful than high ratios. In a "soft-knee" compressor, mostly low ratios are used. No wonder the sound of the compressor is improved! Alas, the price exacted is lost loudness and inconsistent level control. This may be fine for certain applications, but not if you're trying to persuade a wide dynamic range vocal to cut through a heavy instrumental backing, or if you're trying to make anything audible at all times.

What then, **does** make a good-sounding compressor/limiter? Primarily, the dynamic response of the control loop. In the 424A, attack and release times are always "automatic" (i.e., program-controlled); the ATTACK TIME and RELEASE TIME controls merely scale the processes faster or slower without giving up the advantages of the program control.

If a slower attack time is chosen, an automatic gain-riding (AGC) function is achieved. Faster attack times introduce more peak limiting. So, depending on attack time, the 424A can serve as a compressor, limiter, or both simultaneously.

What Makes The 424A So Special?

The Orban 424A has a real compression ratio control for those few times when you want to maintain some **real** dynamic range. However, you will probably be astonished at the "openness" and **apparent** dynamic range available even at the "infinite" ratio. You may be even more astonished when you discover the apparent loudness increase achieved without the usual side effects. Virtually any competent compressor can sound natural if it is quiet enough (i.e., if it doesn't actually work very hard). The **real magic is sounding loud and natural simultaneously, as the 424A does.**

After you've lived with a 424A for a while, we suspect that the dust will build up on the ratio control as you realize that, finally, here is a compressor which doesn't have to be cranked back to a 2:1 or 4:1 ratio (whether manually, by choice, or involuntarily, by means of a "soft knee") to sound good! For those of you who won't give up the "soft knee" no matter what, we've hedged our bets: you'll be pleased to know that at the lower settings of the 424A RATIO control, the ratio increases as more gain reduction is used.

There is also control over the **shape** of the release characteristic. Ordinarily, the release characteristic is "linear." (Recovery, in the absence of program material, proceeds at a constant number of dB per second.)

A switch-selectable "exponential" release shape is also provided for special applications. This forces the release to start slowly and to increase in speed as it proceeds.

While it sounds substantially less natural than "linear" on most program material, it can be useful when gain-riding wide-dynamic range single tracks (like vocalists) where the "open" sound of a slow release time is desired, yet quick gain-riding is necessary to make levels more consistent.

The IDLE GAIN control is a unique and unusually useful new feature. This control sets the gain of the compressor when it is in the "idle" mode. (When the compressor is manually defeated by operation of the DEFEAT switch, or, when it is gated by low-level audio or silence.)

In either case, the gain will move smoothly toward a value specified by the setting of the IDLE GAIN control—quickly after manual compressor defeat, and more slowly under gated conditions.

If the IDLE GAIN control is set close to the average gain reduction, then, upon resumption of ordinary compression, there will be minimum gain change in the VCA, and therefore minimum audible side effects will occur. The feature is extremely useful in preventing noise or tape hiss from being pumped up, and in facilitating smooth manual switching of the COMPRESSOR OPERATE/DEFEAT switch during program. Its effect is usually substantially smoother than that of a conventional noise gate.

The OUTPUT TRIM control can be used to force some peak clipping in the VCA in applications requiring tight control of peak levels (like the protection of a broadcast STL), thus controlling fast peaks without need for extremely fast attack times (which almost invariably cause more audible degradation than a modest amount of overshoot clipping). Conveniently, the peak-reading VCA LEVEL meter not only allows you to adjust the 424A to **clip if you want it to,** but also makes it easy to **avoid clipping entirely** if that is your goal.

About Distortion And VCA's

In any compressor/limiter, the static nonlinearities of the VCA are ordinarily totally overshadowed by the dynamic distortions caused by literally intermodulating the input signal by a rapidly varying control voltage. Sometimes such distortion is heard as additional unwanted spectral components, such as traditional harmonic and IM distortion. At other times, it is perceived as unnatural modulations of the signal peculiar to AGC devices such as "pumping," "hole-punching," "shivering," and a whole bunch more which no one has bothered to name, but which most musically-sensitive people can easily hear!

In the Orban 424A, nonlinear control voltage smoothing results in singularly favorable **dynamic** distortion properties. Our proprietary VCA complements this low dynamic distortion by slewing quickly, having "soft" low-order static distortion which is well below the threshold of audibility (due, in part, to class-A operation), and having noise performance which does not deteriorate as the amount of gain reduction increases.

The Final Coup: A Full-Function De-Esser

Finally, our de-esser. The VCA used in the 424A has **two** gain-control inputs. This means that we can add a **no-compromise** de-ess function which is essentially **independent** of the compressor/limiter. As we have proven in our dedicated de-essers, the optimum attack times and release times are quite different for the compressor/limiter and de-esser functions. In the 424A, each function is independently optimized.

The de-esser section of the 424A sounds about the same as our popular dedicated de-essers. It controls sibilance levels by quick broadband gain reduction when excessive "ess" energy is detected. In this way, any low-frequency IM distortion is reduced along with the "ess," and coloration of the "ess" sounds does not occur. One extra benefit of the de-esser section of the 424A is that it can effectively de-ess sibilant vocals which have already been **mixed** with other program, and, in this application, effectively acts more like a high-frequency limiter.

The only essential difference between the 424A de-esser and the Orban dedicated de-essers is that the 424A lacks the dedicated de-essers' ability to provide constant de-essing regardless of input level. In the 424A, this does not create a problem since it was assumed that the input to the de-esser section would be compressed by the 424A's compressor/limiter and would therefore be at a constant level. LED's indicating NORMAL/HEAVY de-essing allow the SENSITIVITY control of the de-esser to be readily adjusted.

A STEREO COUPLING switch allows you to operate the two channels of the 424A independently or in stereo. In stereo, the channels will typically track within ±0.5dB. Rear-panel coupling terminals allow tracking an unlimited number of channels together.

OPTIMOD-TV:

OPTIMOD-TV Model 8180A is an integrated signal-processing *system* which replaces conventional compressors, limiters, and clippers. It is an adaptation of the OPTIMOD-FM Model 8100A to TV audio, and offers the TV broadcaster the same superb audio quality that has made the 8100A so popular among FM radio broadcasters.

The 8180A is also ideally suited for conditioning signals prior to satellite uplinks, as well as for audio processing in other specialized communications applications where extremely high audio quality is desired, the audio spectrum must be limited to a 15kHz bandwidth, and the channel peak overload point is abrupt and must not be exceeded.

Briefly, the OPTIMOD-TV system performs the following functions:

1. It rides gain over a range of as much as 25dB, compressing dynamic range and compensating for gain riding errors on the part of operators. The amount of dynamic range reduction ordinarily produced is adjustable. When OPTIMOD-TV is operated at its optimum release time setting, gain riding and compression are virtually undetectable because of advanced program-controlled time constants, level-dependent gating, and multiband compression.
2. It controls potential interference to video and/or future stereo services by means of bandwidth-limiting 15kHz lowpass filters incorporating full overshoot compensation. OPTIMOD-TV thus provides extremely tight control over peak modulation, preventing overmodulation and controlling its output spectrum simultaneously.
3. The OPTIMOD-TV compressor is a dual-band design which can be operated with the bands independent of each other ("independent"), or with the bands coupled to ordinarily track each other ("wideband"). When operated in "independent" mode, OPTIMOD-TV can make audio quality more consistent by correcting frequency balances between bass and midrange material. When operated in "wideband" mode, OPTIMOD-TV will preserve frequency balances and will produce an output which sounds like its input.
4. OPTIMOD-TV prevents peak overload and overmodulation due to the effects of the preemphasis curve.

Each part of the OPTIMOD-TV system has been precisely engineered to be compatible with all other parts to achieve optimum performance. The basic OPTIMOD-TV is a stereo unit with active balanced +10dBm outputs. Internal jumpers determine if the output is to be flat or preemphasized, to assure compatibility with any exciter.

In general, OPTIMOD-TV should be fed unprocessed audio. However, additional compression and/or other audio processing may be desirable if applied to individual microphone channels in a live production environment, or to other sources requiring special processing.

OPTIMOD-TV as ordinarily sold is fully equipped for stereo operation in anticipation of the approval of stereo standards. Little would be saved by eliminating components for the second channel, and future conversion would be somewhat inconvenient. In addition, the second channel provides a certain measure of redundancy and permits utilization of card-swapping techniques to diagnose problems.

For stereo, an external stereo generator must be used.

An optional accessory port is available to interface the noise reduction encoder which will probably be required as a part of any multiplex TV stereo system.

Split Configuration: An alternate, dual-chassis system configuration permits the Dual-Band Compressor to be operated separately from the remainder of the circuitry. This permits placing the Dual-Band Compressor at the studio side of the STL (telephone line, dual microwave, or FM subcarrier on a video STL) to protect the STL from overmodulation.

This configuration consists of an OPTIMOD-TV Model 8180A in conjunction with a Model 8180A/ST Accessory Chassis and two jumper cards. Cards #3, #4, and #5 are removed from the 8180A (Main) Chassis and placed in the Accessory Chassis. Card slots #3 and #4 in the Main Chassis are then fitted with the jumper cards.

If the gain between the output of the Accessory Chassis and the input of the Main Chassis cannot be maintained within a 0.75dB window, we recommend using the full single-chassis OPTIMOD-TV at the transmitter, and using an Orban Compressor/Limiter at the studio side of the STL to protect against STL overload.

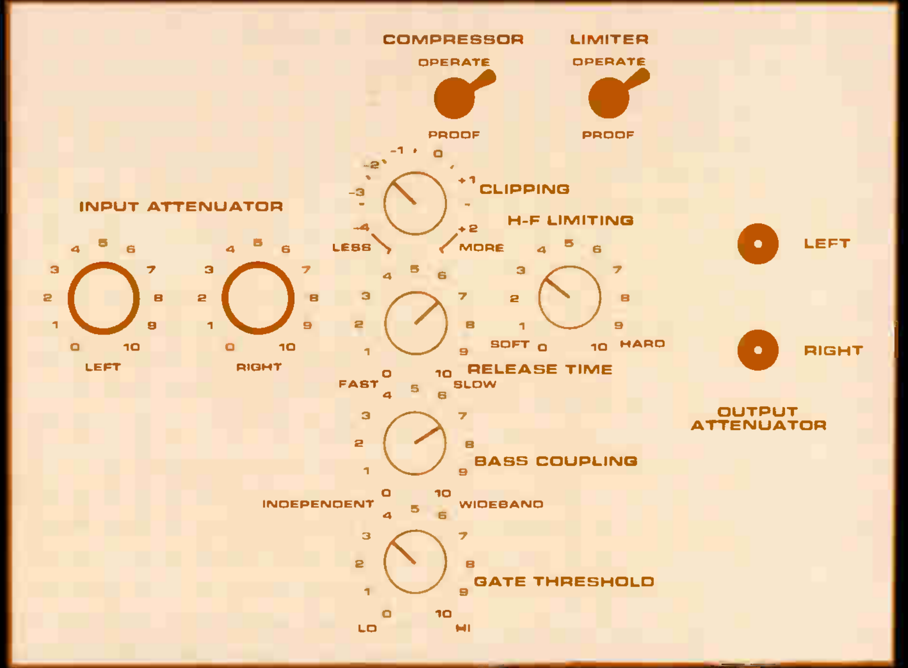

FIGURE 1: RECOMMENDED INITIAL CONTROL SETTINGS

Specifications

FREQUENCY RESPONSE
(System in PROOF mode)

Follows standard 75us preemphasis curve ±0.75dB, 50-15,000 Hz. 50us preemphasis available on special order. Deemphasis jumper on line amplifier card permits flat output ±0.75dB 50-15,000Hz for use with external preemphasis. All preemphasis networks include a fourth-order lowpass filter and fourth-order phase corrector prior to the high-frequency limiter and clipper to prevent these elements from processing out-of-band program material and to minimize overshoot, thus minimizing the amount of high-frequency limiting and clipping.

INPUT CONDITIONING

Highpass Filter: Third-order Chebychev with 30Hz cutoff and 0.5dB passband ripple. Down 0.5dB at 30Hz; 10.5dB at 20Hz; 31.5dB at 10Hz. Protects against infrasonic destabilization of certain exciters' AFC's, as well as infrasonic gain modulation in the compressor.

Phase Scrambler: Allpass network makes peaks more symmetrical to best utilize the symmetrical peak overload characteristics of the FM medium.

NOISE

−75dB below 100% modulation, 50-15,000 Hz maximum; −81dB typical.

TOTAL SYSTEM DISTORTION
(PROOF Mode; deemphasized; 100% Modulation)

Less than 0.05% THD, 50-15,000Hz (0.02% typical); less than 0.05% SMPTE Intermodulation Distortion (60/7000Hz; 4:1).

NOTE

"THD" is defined as the root-sum-square (R.S.S.) sum of all harmonics, 50-30,000Hz. Noise (which is unavoidably included in the reading on a typical THD analyzer) is specifically excluded from this specification. The R.S.S. sum of all harmonics plus noise may measure as high as 0.07% at 15kHz.

"MASTER" BAND COMPRESSOR CHARACTERISTICS

Attack Time: approximately 1 ms

Release Time: program-controlled—varies according to program dynamics and amount of gain reduction (see text). Process can be scaled fast or slow by means of continuously variable RELEASE TIME control. Employs delayed release for distortion reduction.

Total Harmonic Distortion (measured at VCA output, OPERATE mode, RELEASE TIME control centered): Less than 0.1%, 200-15,000Hz, +10 to −15dB gain reduction.

Available Gain Reduction: 25dB

Metering: Three dB-linear edgewise-reading gain reduction meters—
MASTER is true peak-reading with electronic acceleration and peak-hold (+10 to −15dB);
COMPRESSOR indicates slow compression component of gain reduction (+10 to −15dB)
LIMITER indicates fast peak limiting component of gain reduction (0-5dB)

Gain Control Element: True VCA. Proprietary Class-A design eliminates crossover notch distortion, modulation noise, and slewrate limiting found in competitive Class-AB designs.

"BASS" BAND COMPRESSOR CHARACTERISTICS

Attack Time: program-controlled; not adjustable

Release Time: program-controlled; not adjustable. Incorporates delayed-release distortion reduction.

Total Harmonic Distortion: (at VCA output, OPERATE mode): Less than 0.1% THD, 50-200Hz, +10 to −20dB gain reduction.

Available Gain Reduction: 30dB

Metering: single dB-linear edgewise-reading gain reduction meter (+10 to −20dB).

Gain Reduction Element: Proprietary Class-A true VCA

Basscoupling: (U.S. patent #4,249,042): Enables gain of "Bass" band to track gain of "Master" band to any degree, from identical tracking to fully independent operation. Adjustable with BASS COUPLING control.

CROSSOVER CHARACTERISTICS

Control: 6dB/octave @200Hz;

Program: 12dB/octave @200Hz in unique "distributed crossover" configuration (U.S. patent #4,249,042)

HIGH FREQUENCY LIMITER CHARACTERISTICS

Attack Time: approximately 5ms

Release Time: approximately 20ms. Delayed release included for distortion reduction.

Mode: Left and right channels operate independently to avoid high frequencies in one channel causing audible timbre modulation of opposite channel.

Control Element: Junction FET

Metering: Two LED's indicate HF limiting in L and R channels.

Threshold of HF Limiting: User-adjustable over 3dB range to meet format requirements.

FM "SMART CLIPPER" OUTPUT PROCESSOR CHARACTERISTICS

Nominal Bandwidth: 15.4kHz

Distortion Cancellation: Clipping distortion (below overshoot compensator threshold) cancelled better than 30dB (40dB typical), 0-2200Hz (U.S patent #4,208,548).

Delay Correction: Fourth-order allpass

Amount of Clipping: User-adjustable over 6dB range to match program format requirements.

FREQUENCY-CONTOURED SIDECHAIN (FCS) OVERSHOOT COMPENSATOR CHARACTERISTICS (patent pending)

System Overshoot: The FCS circuit is best thought of as a "bandlimited safety clipper." It operates like a hard clipper, but does not produce out-of-band frequency components as a simple hard clipper would. Because the audio processing will sometimes limit steady-state material with high average energy (like sinewaves) or with very little high-frequency energy to levels below the threshold of clipping, it is difficult to state a clear and meaningful specification for the *system* overshoot performance of the FCS circuit.
The FCS circuit is followed by a safety clipper. The overshoot specification could be slightly improved if this safety clipper were set up to clip more frequently. However, the system is aligned at the factory such that the safety clipper is almost *never* active, thus fully preserving the bandlimiting provided by the FCS circuit. With this safety clipper alignment, the peak modulation will be controlled ±3.5% on arbitrary waveforms clipped to any degree by the FCS circuit (acting as a bandlimited safety clipper); peak modulation will not exceed this level on other material. With typical program material, peak modulation uncertainty is less than 2%.

Sinewave Modulation Ability: 93% modulation (i.e., 0.6%dB below maximum overshoot level) at all sinewave frequencies, assuming sinewaves are applied to FCS input.

Dynamic Separation: better than 45dB

Difference-Frequency Intermodulation: FCS circuit causes no more audible IM (such as sibilance splatter) than would a simple hard clipper clipping to the same depth. The entire OPTIMOD-TV processing system is specifically configured to prevent the FCS circuit from audibly degrading the difference-frequency distortion-cancellation properties of the earlier FM "Smart Clipper."

SYSTEM SEPARATION

Greater than 50dB, 50-15,000Hz; 60dB typical

INPUT

Impedance: greater than 10K ohms, electronically balanced by means of true instrumentation amplifier. Requires balanced source.

Common Mode Rejection: Greater than 60dB @60Hz

Sensitivity: −10dBm produces 10dB "Master" Band gain reduction @1kHz. Removal of internal 20dB pad permits −30dBm to produce same effect.

Connector: Cinch-Jones 140-style barrier strip (#5 screw).

OUTPUT

Source Impedance: 370 ohms, independent of OUTPUT ATTEN setting, balanced.

Level: variable −infinity dBm to greater than +20dBm by means of 15-turn OUTPUT ATTEN controls.

Connector: Cinch-Jones 140-style barrier strip (#5 screw). RF suppressed.

AUXILIARY INPUT/OUTPUT
(for Test use only)

Provides L and R lowpass filter output or L and R line amplifier input depending upon setting of rear-apron NORMAL/TEST switch. Connectors are RCA phono-type, unbalanced.

OPERATING CONTROLS

VU Meter Selector: switches ASA-standard VU meter to read:
L or R Input Level
L or R Compressor Output
L or R Filter Out
L or R Line Amplifier Output
±15 V Power Supply Voltages

SETUP CONTROLS (front-panel, behind lockable swing-down door—see Fig. 1)

Compressor:
- Left and Right Input Attenuators
- "Master" Band Release Time
- Gate Threshold
- Bass Coupling
- Clipping
- High-Frequency Limiter Threshold

General:
- Left and Right Output Attenuators
- PROOF/OPERATE Switches (to defeat gain reduction, HF limiting, clipping, and gating)
- Power ON/OFF
- 115V/230V Selector Switch

POWER REQUIREMENT

115/230VAC, ±15%, 50-60Hz, approx. 19VA. IEC mains connector with detachable 3-wire "U-Ground" power cord supplied. Leakage to chassis less than 0.5mA. AC is RF-suppressed.

DIMENSIONS

19" (48.3cm)W × 7" (17.8cm)H × 12.5" (31.2cm)D—4 rack units

ENVIRONMENTAL

Operating Temperature Range: 0-50 degrees C (32-122 degrees F).

Humidity: 0-95% R.H., non-condensing

WARRANTY

One year, parts and labor. Subject to limitations set forth in our Standard Warranty.

All specifications subject to change without notice.

OPTIMOD-TV

Simplified System Description: OPTIMOD-TV consists of six basic blocks:

1. Input Conditioning Filter: This consists of an allpass phase scrambler to make peaks more symmetrical (thus reducing clipping distortion and listener fatique), and a 30Hz 18dB/octave highpass filter to prevent subsonic information from disturbing the operation of the audio processing or exciters' AFC's. Even if an AFC doesn't unlock, it can attempt to "track" subsonic information, producing IM distortion. [The 30Hz highpass filter can be defeated (although we have purposely made it slightly inconvenient to do so); the phase scrambler is an essential part of the system and is non-defeatable.]

2. Dual-Band Compressor: This consists of two compressors in parallel: "Bass" which processes audio below 200Hz (12dB/octave crossover), and "Master" which processes above 200Hz. A BASS COUPLING control adjustable by the user determines if the two bands will operate discriminately ("independent" mode), or if the "Bass" band will be forced to track the "Master" band ("wideband" mode), thus preserving frequency balances. Intermediate bass coupling settings are also available.

Even in "wideband" mode, the bass control loop is still active. Heavy bass will cause a momentary reduction in the gain of the "Bass" band rather than forcing gain reduction of the entire signal (as in a true wideband system), thus avoiding pumping.

Time constants and other parameters of the Dual-Band Compressor have been adjusted so that the summed and preemphasized output of the two bands can be directly applied to the FM "Smart Clipper"(TM). No further gain reduction is required for distortion control, and maximum naturalness is preserved.

The release time of the "Master" band *only* is adjusted with the RELEASE TIME control, thus permitting loudness/fatigue tradeoffs according to your needs.

Gain reduction in both "Master" and "Bass" compressors is metered by edgewise-reading meters calibrated with a dB-linear scale. To indicate that the normal gain reduction is 10dB, this point has been calibrated as "0dB gain reduction" on the meters, with the scale extending from +10 to −15dB gain reduction.

A GATING function is provided which prevents noise rush-up during program pauses (particularly with noisy 16mm optical sound tracks) and makes the 25dB gain reduction range usable. The GATING function is designed such that the gain does not get "stuck" forever in the 0 to 15dB gain reduction region, so low-level program material is *eventually* increased in level. Since gain recovery is slow in GATED mode, the gradual increase in level is essentially imperceptible.

3. Preemphasis And High Frequency Limiter: The summed outputs of the two compressors are applied to a phase corrector, 24dB/octave 15kHz lowpass filter, preemphasis network, and high-frequency limiter. The purpose of the lowpass filter is to prevent out-of-band components from affecting the operation of the high-frequency limiter and to avoid intermodulation between out-of-band frequency components and in-band frequency components in the clipper. Phase correction reduces the peak level increase caused by filter ringing and preemphasis to the theoretical minimum, thus reducing the amount of clipping.

The high-frequency limiter is controlled by high frequencies *only* (rather than by the peak level of the preemphasized signal), eliminating any possibility of modulation of high frequency content by low frequency material.

The threshold of limiting of the high-frequency limiter is user-adjustable over a 3dB range, permitting brightness and high frequency distortion to be traded off according to your needs. Because the patented FM "Smart Clipper" incorporates IM distortion cancellation, substantially more clipping can be accomplished without objectionable distortion than in conventional systems, and significantly improved high frequency power handling capability is achieved.

4. FM "Smart Clipper" (TM): The "Smart Clipper" provides the peak limiting function, and contains filters to assure that the clipping does not introduce out-of-band frequency components above 19kHz which could interface with the video or cause aliasing distortion in future stereo operation.

The output of the high frequency limiter is applied to a clipper with automatically-varying threshold. This clipper performs the basic peak limiting function. The output of the clipper is subtracted from its input, thus deriving the distortion *added* by the clipper. This distortion component is lowpass filtered at 2.2kHz (the knee of the 75us preemphasis curve), and then added to the clipped signal. This "smoothing signal" cancels all clipper-induced distortion below 2.2kHz by 30dB or more, and is particularly effective in eliminating the effects of high-frequency IM, such as sibilance splatter.

The 2.2kHz distortion-cancelling lowpass filter has a time delay. To assure proper distortion cancellation, the main clipped signal must be delayed by an equal amount before the main signal and distortion-cancelling signal are added. The main signal is delayed by a phase-corrected 15kHz lowpass filter, which also removes all out-of-band harmonics caused by the clipping process.

5. Frequency-Contoured Sidechain (FCS) Overshoot Corrector: The output of the "Smart Clipper" contains overshoots due to the addition of the distortion-cancelling signal, and to unavoidable overshoots in the 15kHz filter. These overshoots must be eliminated without adding out-of-band frequency components. This is done in the FCS Overshoot Corrector.

The FCS circuit first derives that part of the signal exceeding the 100% modulation point by means of a "center-clipper." If these overshoots were then *subtracted* from the input signal, the overshoots would be cancelled—in fact, doing so would be equivalent to simple clipping. Unfortunately, this can't be done because the overshoots contain out-of-band frequency components.

The overshoots are therefore lowpass filtered to eliminate out-of-band components. If the overshoot filter had a *flat* response to its cutoff frequency, this filtering action would reduce the amplitude of high-frequency overshoots (by removing out-of-band harmonics which make the overshoots "spikey"). This would result in incomplete cancellation of the overshoots after subtraction. The overshoot filter is therefore designed to have a *rising* response at 15kHz, effectively increasing the gain of the *fundamentals* of the higher-frequency overshoots and compensating for the fact that their harmonics have been removed. The overshoot extractor and this filter are the "Frequency-Contoured Sidechain."

The overshoot filter has phase shift. Phase shift networks are therefore included in the main path to make sure that the overshoot subtraction process works correctly, and that the overall FCS system has constant time delay.

The rising response of the overshoot filter means that essentially no extra subtraction gain (compared to the system operated *without* the filter as a simple differential clipper) is required. Any low frequency IM introduced by the FCS circuit is therefore no worse than the low-frequency IM caused by a simple clipper.

Because the FCS circuit is an instantaneous system and uses no gain reduction or dynamic filtering, it causes neither pumping nor dulling of program material.

6. Output Amplifier/Deemphasis: At the output of the Overshoot Corrector, the signal is peak-controlled and preemphasized. This output is applied to a balanced transformerless line amplifier with strappable deemphasis.

OPTIMOD-TV's outstanding peak-level control capabilities may be fully exploited by defeating exciter preemphasis and applying the pre-emphasized signal from OPTIMOD-TV to the flat exciter. However, in some systems it is extremely inconvenient to defeat the exciter's preemphasis, and the exciter must be supplied with a "flat" (i.e., deemphasized) signal from OPTIMOD-TV.

The output of the line amplifiers are interfaced to the outside world through non-overshooting RFI filters effective from approximately 500kHz to 1GHz.

VU Meter: The front-panel VU meter can monitor the audio level at several different points in the circuitry as selected by the METER switch, and significantly facilitates maintenance and troubleshooting.

Summary: OPTIMOD-TV is an integrated "system approach" to:

- ☐ ride gain
- ☐ perform compression as desired
- ☐ control peaks by high-frequency limiting, distortion-cancelling clipping, and bandlimited overshoot correction.

This optimizes technical parameters to their practical limit while producing a sound at the viewer's ear which is perceived as natural, pleasant, and free from the processing artifacts that often plague other signal processing approaches.

orban

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Broadcasting

The aggressive compression and limiting employed by many broadcasters in an effort to win audience share may result in a substantial exaggeration of announcer sibilance. In addition, there are some "problem announcers" with unnaturally sibilant voices whose sibilance may be unacceptably aggravated by even moderate audio processing and/or the choice of microphones.

Such problems can be solved with the 536A. It can be used beneficially by itself or in a complete signal processing chain for voice only, aided by equalization (like the Orban 622 or 672A parametric equalizers). The extensive RFI protection means easy installation in the broadcast environment.

The 536A can also be used in TV newsrooms and production facilities to combat the sibilance problems often encountered when using "tie-tac" mics. Many female announcers' voices are highly sibilant; the 536A can add a more naturally listenable quality to these voices.

Summary

Virtually everyone involved in professional sound needs de-essing—often at a moment's notice! Either you must deal with at least one "problem voice" which requires de-essing to reach contemporary standards of recorded quality, or sibilance buildup problems will hold you back in your quest to transcend the merely adequate and achieve the truly excellent.

If you recognize either problem, the Orban 536A is your ideal solution. It's clean. . . it's quiet. . . and it's **simple.** It's a **problem solver,** not a problem causer. See your dealer about the new 536A two-channel, rack-mounted de-esser from Orban—the de-essing expert.

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Specifications

Frequency Response: ±0.25dB, 20-20,000Hz

Total Harmonic Distortion: (de-essing defeated): <0.025%, 20-20,000Hz, @ +24dBm

Total Harmonic Distortion: (de-essing in): <0.5% @6kHz

Output Noise: (20kHz bandwidth): < –75dBm. Dynamic range from noise floor to clipping exceeds 100dB.

Input Level Variation for Constant De-essing: >15dB

Input Characteristics:
- **Impedance:** >10,000 ohms, active-balanced bridging
- **Level:** –10 or +4dBm (strappable)

Gain: +20dB or +6dB (Dependent on input strap; referred to balanced output. Referred to unbalanced output, gains are 6dB lower.)

Output Characteristics:
- **Impedance:** Approximately 100 ohms, active-balanced-to-ground. Fully-floating transformer output optional. Unbalanced output available from either output to ground.
- **Level:** Drive capability into 600 ohms exceeds +25dBm, 20-20,000 Hz

Crosstalk: < –80dB

Attack Time: approximately 1 ms

Recovery Time: approximately 10 ms

Variable-Gain Element: junction field-effect transistor

Indicators:
- Two-element LED gain reduction meter;
- LED OVERLOAD indicator
- LED POWER ON indicator

Operating Controls:
- THRESHOLD control (each channel)
- DE-ESSING IN/OUT switch (each channel)
- POWER ON/OFF switch

Connectors: All inputs and outputs appear on Jones 140-Y-type barrier strip (#5 screw). Chassis is punched for optional installation of paralleling XLR-type input and output connectors.

Power Requirements: 115-230 volts AC, ±10%, 50-60Hz, approx. 6 watts. Captive power cord with "U-Ground" plug to United States standards.

Size: 19" (48.3cm) wide x 1¾" (4.45cm) high x 5¾" (14.6cm) deep

Weight:
- Shipping: 7 lbs. (5.2 kg)
- Net (without options): 5 lbs. (2.3 kg)

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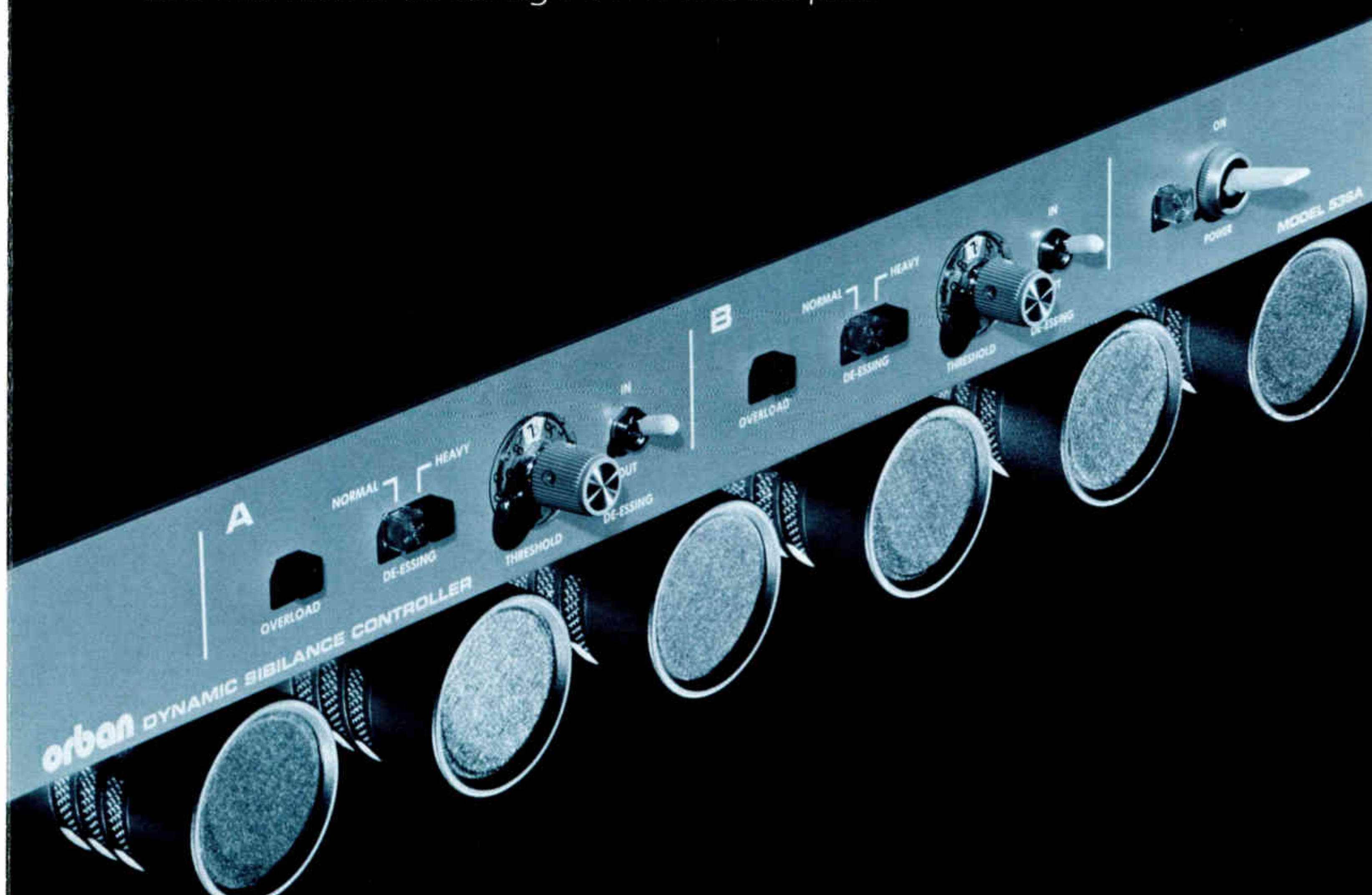
The Orban 536A Dynamic Sibilance Controller

Two channels of de-essing at an affordable price.



Performance Highlights

- ☐Two independent channels
- ☐Effective, inaudible de-essing over a 15dB input range
- ☐Active-balanced input, strappable for +4 or −10dBm
- ☐Active-balanced output with transformer option
- ☐Dual-LED gain reduction metering
- ☐Overload/noise ratio typically 105dB
- ☐Very low distortion
- ☐Effective RF suppression
- ☐Top-quality professional construction
- ☐Cost-effective 19" rack mount package
- ☐Low cost-per-channel

Related Products:
526A Single-Channel Dynamic Sibilance Controller (with switchable mic/line inputs)

ORDER GUIDE

536A Two-Channel Dynamic Sibilance Controller
Options:
RET-22 XLR-type connectors, kit
RET-23 Transformer-coupled outputs(500/600 ohms), kit
Note: RET- kits are factory-installed if requested, or may be ordered later as field-retrofit kits

Description

The new Orban 536A Dynamic Sibilance Controller is an improved, two-channel version of our popular single-channel 526A (but without the mic input). The 526A and its ancestor, the 516EC, are considered to be the industry-standard de-essers and are found in professional audio facilities throughout the world. Now, with the introduction of the 536A, we offer the most cost-effective package available today—two channels of de-essing at an affordable price.

Like its predecessors, the 536A is designed to work effectively on **voice only** in its many professional applications. (Users requiring de-essing of mixed tracks should consider the Orban 422A/424A Gated Compressor/Limiter/De-esser.)

Compared to its competitors, the 536A offers vastly simpler setup, improved noise and distortion performance, and no emphasis of residual IM distortion while de-essing is occurring.

The 536A is active-balanced at both inputs and outputs. For applications requiring a fully-floating output, optional output transformers can be fitted. Like all Orban products, the 536A features RF filtering of input, output, and power leads to help assure trouble-free installation in high RF fields.

The 536A is simplicity itself to operate. There is only one adjustment: a THRESHOLD control. The amount of sibilance energy is limited to a certain **fraction** of the non-sibilant speech energy—even if the input level changes as much as 15dB. The THRESHOLD control determines this fraction. Thus, a consistent balance between the "voiced" and "sibilant" sections of speech is achieved whether or not input levels are well-controlled. This is especially useful with off-mic voices, or in any application (such as motion picture sound) where speech levels vary widely.

A dual-LED display indicates the amount of de-essing which is occurring ("Normal" and "Heavy"), warning of possible misadjustment of the THRESHOLD control. An peak-stretching OVERLOAD indicator warns of clipping anywhere in the circuit.

The IN/OUT switch can be operated at any time without clicks, pops, or gain changes.

Critical parameters, such as attack time, release time, and the frequency characteristics of the control loop, have been preset after extensive listening tests on many voices. Therefore, these parameters can never be misadjusted by the user. Instead, the 536A is simply **there**—ready to do the job quickly and efficiently with minimal hassle.

How It Works

"Esses" are detected by means of a sharply selective filter which effectively discriminates between "ess" frequencies (centered around 6kHz) and lower-frequency vocal components. In addition, the absolute threshold at which de-essing begins to occur is automatically forced to track the peak input level. This threshold is thus a constant **fraction** of the input level, resulting in **constant de-essing action over a wide range of input levels.** (The fraction is determined by the setting of the THRESHOLD control).

When an "ess" attempts to exceed the automatically-varying threshold, the 536A decreases its gain as necessary to reduce the "ess" level back to the threshold. Recovery is so quick that the following vocal sound is not audibly affected. Because the gain of the **entire** channel is reduced (not just the high frequencies), any IM distortion existing on the original recording medium is reduced along with the "ess", and coloration of the "ess" sound does not occur.

Non-linear control voltage smoothing assures that excessive modulation distortion will not occur under de-essing conditions despite the extremely fast (approximately 10ms) release time employed.

The variable-gain device is a junction FET, assuring a gentle overload characteristic and freedom from control-voltage leakage into the signal.

Limitations

There are some limitations: First, using the 536A on voice which has already been mixed with other sounds is often unsuccessful because it can mistake sounds having high frequency content for sibilance, with unpredictable results. If de-essing of mixed vocals and instruments is desired, we recommend that you evaluate our Model 422A (mono) or 424A (stereo) Gated Compressor/Limiter/De-essers which can perform that task.

Another limitation is that, because the 536A uses frequencies of 6kHz and above to control its action, the minimum bandwidth required of the source material is approximately 8kHz. Occasionally, bandwidth problems arise when the "sibilance" is mostly IM distortion caused by tape overload (particularly with cassettes), by use of telephone-quality carbon microphones, or by attempts to de-ess recordings distorted by previous transfer to optical film. If the sibilance exists as IM without significant energy above 6kHz, then the de-esser cannot detect and control it.

The track must also be reasonably free of noise or hiss in the region above 6kHz. Normally these factors do not cause problems except in the case of the lowest-quality program material.

It is known that some dialects of non-English languages can trigger de-essing action on sounds other than sibilance. This is ordinarily not a problem if the THRESHOLD control is set carefully.

Application Ideas

Recording Studio

Current vocal equalization practices in the pop recording industry tend to include large amounts of high frequency boost to increase presence and articulation. These boosts, particularly when combined with limiting or compression, can boost sibilance to unpleasantly high levels. Using the 536A **after** the equalizer and limiter can reduce the sibilance to levels that sound natural and right. With the 536A, equalization and limiting are no longer constrained by the problems of unnatural sibilance levels, and may therefore safely be used to achieve the ultimate artistic goals of the producer and engineer. Use of 536A during recording allows the producer/engineer to get the intimate, "tight to the mic" sound that has become so popular on hit records while substantially reducing the probability of disk overcutting or cassette saturation when the master is transferred to these common consumer media.

The availability of two independent channels of de-essing in the same package means that two different voices can be de-essed simultaneously without interaction, making the 536A extremely cost-effective.

Motion Picture Sound/Video Production

The susceptibility of the variable-area optical soundtrack to high frequency crossmodulation distortion made de-essing compulsory in motion picture recording. The 536A can fulfill this need in a simple, natural, and inaudible way—with solid-state reliability and with the fully balanced input and output interface requirements characteristic of motion picture re-recording theaters.

Similarly, in video post-productions, there are often problems in getting properly-balanced dialogue tracks—particularly when mic placement was compromised by the needs of the video! The 536A allows radical corrective re-equalization of such problem tracks without overload due to sibilance.

Sound Reinforcement

Ultra-close miking is the norm in most amplified music. If substantial 6kHz boost has been applied to increase presence—either by means of an equalizer, or by use of a mic with a built-in "presence peak"—then sibilance and "spitting" can become a real problem. The 536A is a cost-effective solution. Up to two microphone channels can be de-essed simultaneously with minimal setup problems.

Technical Description

The 672A Equalizer consists of a balanced input buffer amplifier, eight main equalization amplifiers connected in series, and tunable lowpass and highpass positive feedback 12dB/octave Butterworth filters. The output of the lowpass filter is buffered to drive 600 ohms, and is available separately. By suitable switch settings, the main output can be made to carry a highpassed signal. Thus the 672A can be used as an equalizer cascaded with a full electronic crossover, or as an equalizer cascaded with lowpass and highpass filters.

Each amplifier in the equalizer section provides equalization for one band only, thus assuring no interaction between bands. The total equalization is simply the sum (in dB) of the equalizations provided by each of the sections.

Peak boost is accomplished by adding the output of a two-pole bandpass resonator to the main signal; reciprocal dip occurs when this resonator is symmetrically connected as a feedback element in the main equalizer amplifier.

The EQ IN/OUT switch bypasses the last seven main amplifiers and defeats equalization in the first amplifier. Gain and signal polarity are equal in the IN and OUT modes. As the BANDWIDTH control is operated, the skirts of the equalization curve move in and out, but the peak gain and peak frequency remain constant. As the EQUALIZATION controls are operated, the frequencies of peak gain remain constant. However, as the TUNING control or EQUALIZATION control (in **dip** mode) are operated, the bandwidth ("Q") will change, because of the simplifications in the "quasi-parametric" bandpass resonator. Careful design has enabled us to produce curves (in **boost** mode only) essentially identical to the desirable "constant-Q" curves provided by our 622B **true parametric** equalizer in its **boost** mode.

The EQUALIZATION controls all produce peaking curves; if shelving curves are desired, they can be approximated by tuning the lowest band to 20Hz and the highest band to 20kHz. The breakpoint of the shelving characteristic is then adjusted with the BANDWIDTH control.

Summary

Many people are now aware of the **power** of parametric equalization: the almost sensual satisfaction of getting the sound really **right.** These same people are also demanding professionals, insisting on inaudible noise and distortion, human engineering, quality "feel", and uncompromising reliability.

Orban is well-known for its line of fine parametric equalizers, like the 622B. Now with the 672A, it brings equalization of the same rigorous quality to applications where it could never before be afforded. The 672A is inexpensive enough to qualify it for serious consideration in applications which would otherwise be given by default to a much less able graphic equalizer.

The 672A rounds out the line of Orban "Professionals' Parametrics." Between the 622B **true parametric** and the 672A **quasi-parametric,** there is an equalizer for virtually every need and budget. The Orban "Professionals' Parametrics" are available at your authorized Orban dealer.

Specifications

All specifications apply when driving 600 ohms or higher impedances. Noise measured on an average-reading meter through a 20-20,000Hz bandpass filter with 18dB/octave Butterworth skirts.

ELECTRICAL

Input:
Impedance, Load (each leg): 100K in parallel with 1000pF, electronically balanced Impedance, Driving: Ideally 600 ohms or less, balanced or unbalanced.
Nominal Input Level: Between −10 and +4dBm
Absolute Overload Point: +26dBm

Output:
Impedance, Source: 47 ohms in parallel with 1000pF, unbalanced (Optional transformer balanced 6000 ohm outputs)
Impedance, Load: Should be 600 ohms or greater—will not ring into any capacitive load
Nominal Output Level: +4dBm
Max. Output Level Before Clipping: greater than ±19 dBm, 20-20,000Hz

Frequency Response:
±0.25dB, 20-20,000Hz: EQ controls set at "0" détents

Available Gain:
+12dB; adjustable to −infinity by means of front-panel GAIN control

Slew Rate:
Varies between 6 and 13V/uS depending upon setting of GAIN control; slewing is symmetrical. Internal bandlimiting assures that slew rate limiting will not occur even with the most severe equalization and program material.

Square Wave Response:
Square wave exhibits no spurious ringing at any output level. The only ringing observable is that theoretically associated with any given equalization curve.

Total Harmonic Distortion:
Less than 0.05%, 20-20,000Hz (+18dBm)

SMPTE Intermodulation Distortion:
Less than 0.05% (+18dBm:60/7000Hz, 4:1)

Noise at Output:
Less than −75dBm (EQ in, filters out, controls centered)

Overload/Noise Ratio:
Better than 113dB for any single bandpass filter, for any settings of TUNING or BANDWIDTH controls.

Equalization Ranges:
±16dB peaking EQ, Reciprocal

Tuning Ranges:
20-60Hz; 40-150Hz; 110-310Hz; 230-750Hz; 480-1900Hz; 1.1-4.5Hz; 2.8-9.0kHz; 5.9-21.kHz. Dials calibrated at ISO preferred frequencies.

"Q" Range:
Greater than 0.5 to 10 for any setting of the TUNING control

Low Pass Filter Section:
Tunable in 2 ranges: 200-2000Hz or 2.0-20kHz, 12dB/octave, (2nd-order Butterworth)

High Pass Filter Section:
Tunable in 2 ranges: 20-200Hz or 200-2000Hz, 12dB/octave, (2nd-order Butterworth)

Overload Indicator:
Lamp lights for 200mS if the instantaneous peak output of any amplifier rises to within 1dB of its clipping point.

Circuit Design:
Active RC realized with FET-input opamps. Line driver employs discrete transistor current booster.

Operating Temperature:
0-50°C

Power Requirements
115/230VAC±10%; 50/60Hz; 6 watts

PHYSICAL

Operating Controls
EQUALIZATION, TUNING, and BANDWIDTH for each of eight bands. TUNING, RANGE (x1; x10), and FILTER IN/OUT for each filter. EQUALIZATION IN/OUT, POWER ON/OFF, and GAIN for entire equalizer.

Panel:
19" x 5¼" (48.3 x 13.3cm): 3 units

Chassis Depth Behind Panel:
5¼" (13.3cm)

Weight:
Net: 8 lbs. (3.6kg); Shipping: 12 lbs (5.4kg)

AC Cord:
3-wire U-ground to USA Standard

Connectors:
140 type barrier strip (#5 screw) plus parallel-wired ¼" 3 ckt. phone jacks (Switchcraft 12B or equal). Holes punched for XLR-type connectors (Switchcraft D3F and D3M or equal)

Circuit Ground:
Available on barrier strip; normally jumpered to chassis.

Specifications subject to change without notice.

orban

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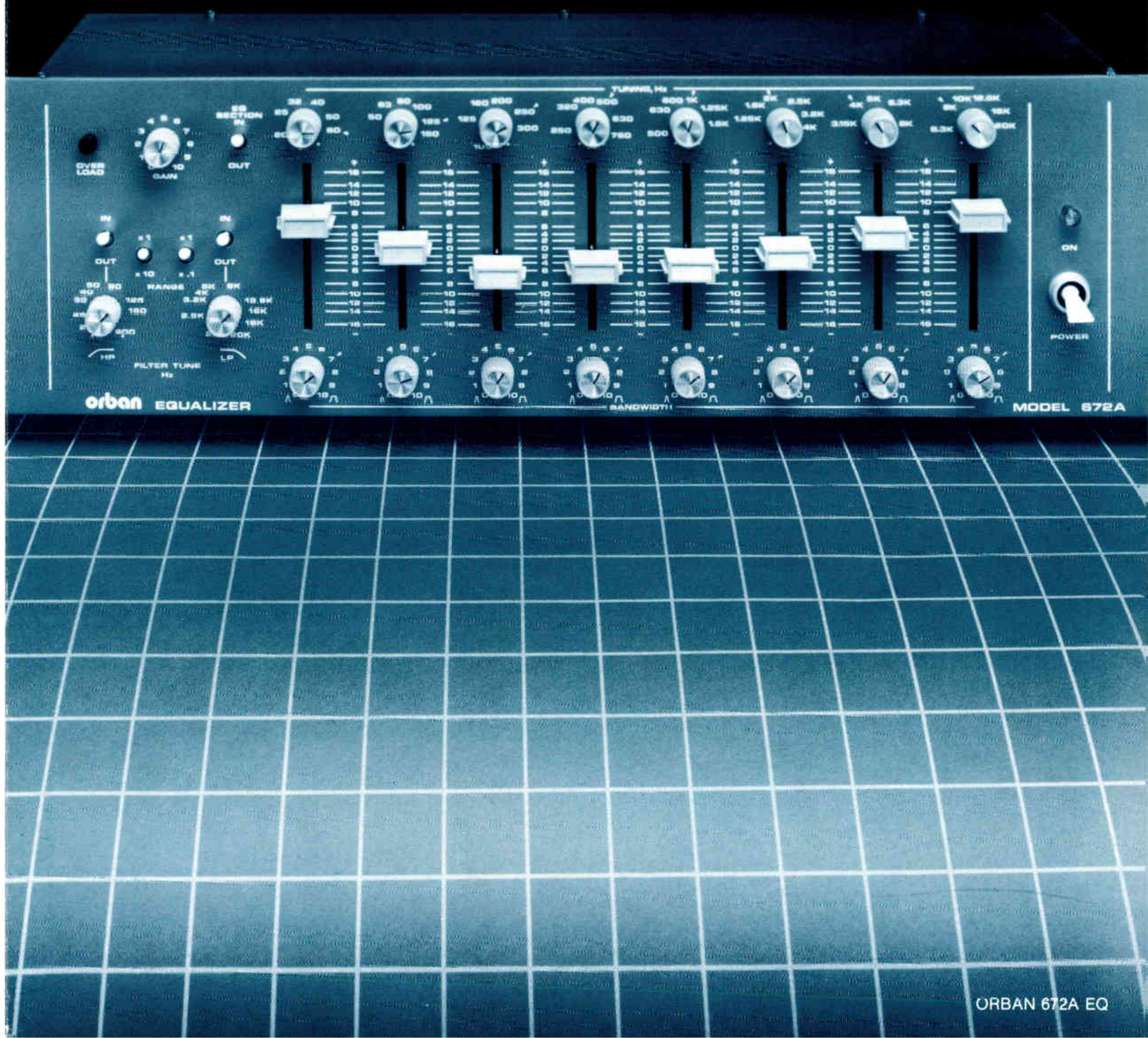
The Orban 672A Equalizer

The versatility of a parametric; The economy of a graphic...
More flexible than either.



PERFORMANCE HIGHLIGHTS

EQ Section

- ☐ Eight bands, each with TUNING and BANDWIDTH control
- ☐ Each band tunes over 3:1 frequency range
- ☐ "Q" typically variable between 0.3 and 20 (center TUNING)
- ☐ ±16dB equalization range
- ☐ EQ controls are long-throw dust-shielded slidepots for good resolution
- ☐ TUNING and BANDWIDTH controls marked with "tics" indicating typical settings
- ☐ Narrowband notching capability ideal for sound reinforcement
- ☐ Bands totally non-interacting

HP/LP Filter Sections

- ☐ Each section continuously tunable over 100:1 range in 2 decades
- ☐ Each section independently switchable
- ☐ 12dB/octave slopes
- ☐ Filters follow graphic section. Separate main/highpass and low-pass outputs allow use as filters or as full electronic crossover

General

- ☐ Very low noise and distortion
- ☐ High slew rate for minimum TIM (SID)
- ☐ Front-panel GAIN control; 12dB gain available
- ☐ "Peak-stretching" overload lamp warns of clipping anywhere in equalizer
- ☐ Active balanced input; unbalanced outputs. Transformer-balanced outputs optional
- ☐ RF suppression on input, output, and power leads
- ☐ 115/230V, 50-60Hz transformer is standard
- ☐ Industrial-grade parts and construction including socketed IC's
- ☐ Highly cost-effective

A stereo version of this product is also available as Model 674A

INTRODUCING THE 672A EQUALIZER

The Orban 672A is a cost-effective, professional, quasi-parametric equalizer with the convenience of graphic-type EQ controls. Wide-range high- and low-pass filters with 12dB/octave Butterworth slopes follow the EQ section for added versatility. The 672A has two outputs, arranged so that these filters can also be used as a fully tunable electronic crossover.

The 672A is a professional product designed to provide a large measure of versatility, convenience, and quality at a very attractive price. While it meets the requirements of the demanding professional, it is also designed and priced to make it understandable and available to the advanced audiophile.

To make the 672A easy to use in situations where its full versatility isn't needed, "tic" marks have been included on the dial calibrations of the TUNING and BANDWIDTH controls. When these controls are set to the tics, the 672A behaves like a standard octave-band graphic equalizer with the eight bands on ISO frequencies from 63 to 8000Hz.

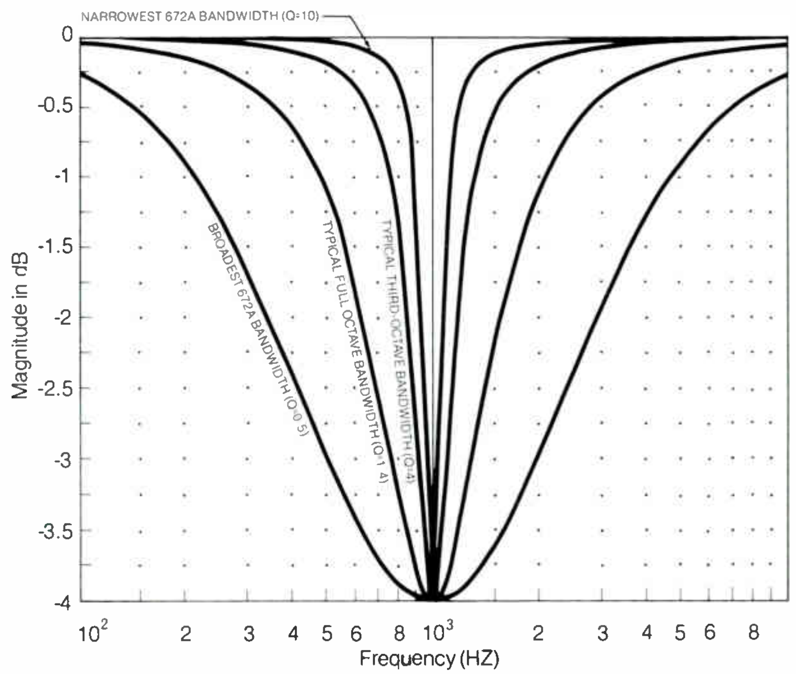
Each feature of the 672A has been thoughtfully chosen and cleverly implemented to make the equalizer a particularly powerful tool in nearly all areas of audio: sound reinforcement, public address, recording studio, broadcasting, motion picture sound, disco, theater. . .

Why "Quasi-Parametric"?

There are two basic types of parametric equalizer: **full-** and **quasi-** parametric. Orban manufactures both types. Both offer far more effective control than other kinds of equalizers, like graphics. Our popular dual-channel 622B is a **full** parametric. This means that you have **totally non-interacting** control over the three fundamental **parameters** of equalization: the **amount** of peak boost or dip (in dB), the **tuning** (the frequency most affected by the equalization), and the **"Q"** (which relates to the sharpness of the EQ curve—the degree to which frequencies on either side of the peak frequency are affected by the equalization). As opposed to our 622B Equalizer, the 672A is **quasi**-parametric. This means that the "Q" changes when you adjust the TUNING and/or EQ controls. Other control adjustments are completely non-interacting: TUNING and EQ do not affect each other.

The other important performance difference between the full-parametric 622B and the new 672A is that the 622B's EQ curves are "**constant-Q**"; the 672A's curves are "reciprocal." "Constant-Q" curves are valuable in that they permit infinite-depth notches to be created; reciprocal curves limit the maximum cut to the same number of dB as the maximum boost. In the case of the 672A, 16dB of cut is available. This is fine for tuning out ring-modes in sound reinforcement systems, but might not be adequate in all circumstances to remove hum or other fixed-frequency interference from a signal. On the other hand, some people prefer reciprocal curves because the boost and cut are mirror images of each other, thus permitting previous equalization to be readily "undone" later. Careful design of the circuitry gives the 672A **in boost mode** a characteristic similar to the 622B's desirable "constant-Q" curve family.

Why did we choose the quasi-parametric technique for the 672A? Because it offers a way to produce a very high quality, stable equalizer at low cost without compromising distortion, noise, accuracy, or reliability.



Various Bandwidth Dipping Curves — 4dB Dip

APPLICATIONS

Sound Reinforcement and Monitor Tuning

There are many ways to use the 672A in sound reinforcement:

1) In an economy biamped installation, replace both the third-octave equalizer **and** the electronic crossover with the 672A. The 672A's narrowband, tunable notches can deal with ring modes more effectively than the third-octave unit could. Use three or four of the 672A bands for narrowband notching; leave the rest for broadband EQ.
2. In a higher budget biamped installation, use the 672A as an electronic crossover plus a narrowband, tunable notch filter for ring mode suppression; incorporate a separate third-octave equalizer to correct the house curve.
3. Use variations of (1) and (2) above **with** an electronic crossover; the 672A's high-pass and lowpass filters can then be used to roll off the frequency response of the system in a controlled manner.
4) In a **non-biamped** system (like a stage monitor), use the 672A to equalize the monitor, and use its filters to restrict response in the extreme high and low frequencies.
5) Use the 672A as a **partial** electronic crossover plus equalizer/filter by devoting one channel of 672A equalization to each driver; one filter is required to perform the crossover function; the other can be used for its normal highpass (or lowpass) function.

In all cases, the BANDWIDTH control can be adjusted to make the totally non-interacting (series-connected) bands "combine" —a most desirable characteristic in sound reinforcement.

Any way you cut it, the 672A's economy and extraordinary versatility make it one of the sound reinforcement practitioner's most useful tools.

Recording Studios

Every recording studio needs a few channels of 672A equalization to handle the tough chores that the internal console equalizers can't deal with. Patch that problem track through a 672A: its fine-tuning ability lets you clean up the track far more effectively than you could with a graphic or "three knob" console equalizer. Use the tunable filters to help eliminate rumble, cymbal splash, kick drum leakage—you name it!

The 672A is also an ideal adjunct to an electronic music synthesizer—you can create high "Q" formants and shape the spectrum so that the sound comes alive.

If you need to correct the equalization of a finished track because of second thoughts after the mix, the 672A can create the finishing touches as no ordinary equalizer can. It's better than a third-octave graphic, because the 672A can generate broad, non-ringing boosts, whereas the graphic is much more colored and ringy.

Motion Picture Sound and Video Sweetening

The 672A is an ideal replacement for the graphic equalizers ordinarily used for dialogue equalization in motion picture sound. Set the TUNING and BANDWIDTH controls to the "tics" on the panel, and you get the equivalent of a familiar, easy-to-operate graphic. But when you **need** the extra control and flexibility—such as notching out the extraneous sounds that always seem to plague location recordings—that power is there instantly, without patching or the use of external dip filters. The high and lowpass filters are invaluable for cleaning up noise and rumble without affecting dialogue—and without using up EQ channels to try to achieve filtering. In addition, many "effects" (such as telephone, pocket radio, or "old time" recordings) can be easily created with the 672A alone.

In addition, the 672A can be used to equalize the "B-chain" in the re-recording theater to the acoustic response specifications of the studio. The lowpass filter can effectively simulate the "Academy Rolloff" or its current modifications.

Broadcasting

Use the 672A in the production studio to enhance the announce mike, and to create special production effects that make your station stand out among its competitors. Meanwhile, another 672A can be quietly and efficiently equalizing the program line for maximum punch and brightness on the air. Use the 672A to equalize phone or remote lines for flat response—it's much more versatile than the standard phone company equalizers. In the main studio, use it on the announce mike channel to equalize for maximum presence, and also to notch out sounds like mechanical hum from cart machine motors or air conditioning noise. Whatever your application, the 672A's RF suppression and optional output transformer mean problem-free installation in high-RF environments.

Dance Bars

The 672A is an excellent dance bar equalizer. The sound contractor installing the system can offer the management exactly the sound desired—including solid, punchy bass free from muddiness and boom—and an aggressive, sizzling top free from the ringing and coloration typical of a full-octave graphic equalizer. The eight bands permit substantial work to be done in flattening out undesired response deviations in the upper bass and midrange. Narrowband notches can even deal with the difficult resonances sometimes encountered in high-efficiency horn-type loudspeakers. In biamped installations, use the separate lowpass and highpass filter outputs as a complete electronic crossover. No other crossover is necessary.

The 672A costs a bit more than an octave-type graphic. But, unlike a graphic, it really **solves** the problems.

The defeatable silence gate (controlled by the front-panel GATE THRESHOLD control) avoids noise rush-up during near-silent passages by radically slowing the release process, thereby forcing the VCA gain to freeze and then to move very slowly toward the average of the last 30 seconds of gain reduction. (This is an automatic "idle gain" function, similar to the manually set IDLE GAIN on our 424A Gated Compressor/Limiter/De-Esser.)

VCA noise in the gain reduction circuitry (typically −90dB below the VCA output clipping point, 20-20kHz unweighted) stays virtually constant as the VCA gain changes. This results in very low noise under real-life conditions. The VCA used is a proprietary Orban Class-A VCA with very low static distortion.

Summary

The Orban Model 464A Co-Operator provides an unprecedented combination of versatility, audio quality, and ease-of-use. Like any such device, specifications and brochures can't really describe what it sounds like or how it feels to use it. We invite you to try the Co-Operator for yourself. We think you will find it to be an indispensible assistant.

Other Orban Level Control Devices

Model 422A (mono)/ 424A (stereo) Gated Compressor/ Limiter/De-Esser: A very smooth-sounding, full-featured, gated compressor/limiter/de-esser. Provides manual control of gate threshold, attack time, release time, compression ratio, idle gain, and de-essing threshold. The 422A/424A is the best choice where the need for additional versatility justifies the time and skill necessary to set the many manual controls, and where HF limiting is not required.

Model 412A (mono)/ 414A (stereo) Compressor/Limiter: A low-cost version of the 422A/424A with an added threshold control (but without gating or de-esser). Ideal for sound reinforcement applications.

Specifications

Performance

INPUT

Impedance: >10K ohms, active balanced, EMI-suppressed.

Operating level: Usable with nominal levels from −10dBm to +8dBm.

OUTPUT

Impedance: 93 ohms, electronically balanced and floating to simulate true transformer output. Minimum load impedance is 600 ohms. Output can be unbalanced by grounding one output terminal.

Level: Front-panel controls permit use with −10dBm to +8dBm systems. Output clipping level is >+20dBm into 600 ohms.

SYSTEM

Frequency response (20–20,000Hz): ±0.25dB below leveler, compressor, and high-frequency limiter thresholds.

RMS noise (20–20,000Hz): >85dB (90dB typical) below output clipping threshold with high-frequency limiter strapped for flat output.

Interchannel crosstalk: Better than −60dB at 15kHz (−67dB typical), falling at 6dB/octave below 15kHz.

Circuitry

LEVELER/COMPRESSOR

Attack time: Approximately 200ms for leveler, 5ms for compressor; program-dependent.

Release time: Adjustable between approximately 1dB/sec and 5dB/sec; program-dependent. Below 10dB gain reduction, either a constant dB/sec release rate (HARD RELEASE SHAPE) or an exponentially declining rate (SOFT RELEASE SHAPE) may be selected.

Compression ratio: >20:1 (static); program-dependent (dynamic).

Range of gain reduction: 25dB.

Interchannel tracking: ±0.5dB (with MODE button set to STEREO).

Total harmonic distortion: <0.05% at 1kHz (with RELEASE TIME controls centered and 15dB gain reduction). Typically <0.1% at 20Hz, <0.03% from 100–2,000Hz, <0.05% from 2,000–10,000Hz, and <0.1% from 10,000–20,000Hz.

SMPTE intermodulation distortion: <0.05% (60/7,000Hz 4:1 with 15dB gain reduction).

Gain reduction element: Class-A proprietary VCA.

HIGH-FREQUENCY LIMITER

Pre-emphasis: Six switch-selectable 6dB/octave pre-emphasis curves: 25, 37.5, 50, 75, 100, and 150 μs. Can be strapped for flat or pre-emphasized output. A defeatable peak clipper can enforce an absolute peak ceiling on the (pre-emphasized) output of the HF limiter.

Response: The high-frequency limiting threshold and attack time have been set so that no audible distortion is produced with dynamic program material that has been processed by the leveler/ compressor and peak clipper. Because these settings have taken into account the peak-to-average ratio of the leveler/ compressor's output, it is not possible to specify the high-frequency limiter's response to test tones with simple, meaningful numbers.

Total harmonic distortion: The high-frequency limiter/clipper will add no more than 0.02% THD to sinewave test tones that have been processed by the leveler/compressor.

Release time: Approximately 30ms, program-dependent.

Interchannel coupling: Each channel's high-frequency limiter operates independently at all times (the fast release times do not disturb the stability of the stereo image).

Gain reduction element: Junction FET.

HF limiting curve: Shelving, 6dB/ octave.

CONTROLS & METERS

Buttons: "Wink-eye" type—end of buttons turn white when pressed in.

Meters: Four 10-segment LED bargraph displays show gain reduction and peak output level for each channel.

Indicators: Four LEDs light to show operation of gating or high-frequency limiting.

Installation

Power requirements: 115/230 volts AC ±10%, 50–60Hz, 16VA. Three-wire "U-Ground" power cord (to USA standards) attached. EMI-suppressed.

Dimensions: 19.0″ (48.3cm) wide, 9.625″ (24.5cm) deep, 1.75″ (4.5cm) high.

Operating temperature range: 32–113°F (0–45°C)

Input/Output connections: Barrier strip (#5 screws) in parallel with ¼″ balanced phone jacks. Chassis is punched to accept optional XLR connectors.

Warranty: One year, parts and labor. Subject to the limitations set forth in Orban's Standard Warranty Agreement.

orban

Orban Associates Inc., 645 Bryant Street, San Francisco, CA 94107
(415) 957-1067 Telex: 17-1480

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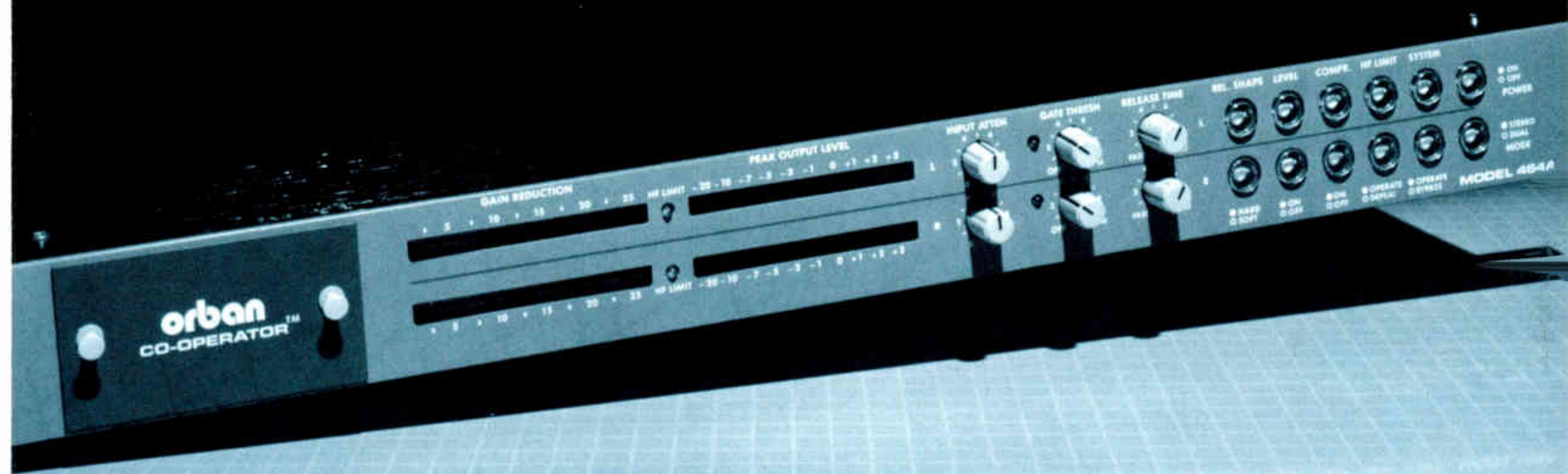
The Orban 464A Co-Operator™

An Integrated, Easy-to-Use
Gated Stereo Leveler / Compressor / HF Limiter / Peak Clipper



Features:

- ☐ A smooth leveler function provides transparent gain riding—without long-term distortion-producing overshoots.
- ☐ A faster "compression" function can be switched in to provide additional transient overshoot protection for material with abrupt level changes or unusually high peak-to-average ratios. The compression function is available without leveling for applications that require safety limiting only.
- ☐ Release time and release shape are adjustable to optimize processing for music or voice.
- ☐ A defeatable "silence gate" prevents noise rush-up, holes, pumping, and breathing by inhibiting sudden gain increases once the signal level falls below a preset threshold (during pauses or low-level program material).
- ☐ Six switchable HF limiter pre-emphasis/de-emphasis curves (25 to 150μs) allow the HF limiting to react optimally for the medium or device being protected and guard against excessive sibilance.
- ☐ A defeatable clipper follows the HF limiter for absolute peak protection.
- ☐ Switchable for stereo-tracking or independent two-channel operation.
- ☐ The least-used controls are concealed behind a security panel to avoid confusing non-technical operators, and to permit tamper-proof calibration.
- ☐ Two LED bargraphs per channel simultaneously display gain reduction and peak output level.
- ☐ Output level meter can be calibrated to match the overload point of the device being driven.
- ☐ Balanced, floating inputs and outputs are EMI-suppressed.
- ☐ 25dB gain reduction range is achieved with a low-distortion, Class-A VCA.
- ☐ Two channels in a space-saving, rugged, all-metal, 1¾" package.
- ☐ Hard-wired bypass switch is included.

Your Assistant Operator

The Orban Co-Operator is a friendly, automatic "assistant operator" that smoothly and unobtrusively rides gain and limits peaks.

After audio quality, ease of use was the highest priority in the development of the Co-Operator. The operator need only be concerned with three controls:

Input Attenuator: Determines the amount of gain reduction.

Gate Threshold: Determines the level below which ordinary automatic gain control (AGC) action is "frozen" (to prevent noise rush-up during pauses and low-level program).

Release Time: Speeds or slows the program-controlled release time to match the processing to the "flavor" of the audio being processed.

The operator can also select the gain reduction recovery rate with the RELEASE SHAPE control. The constant dB/second HARD recovery rate is intended for use with single tracks and live voice; the SOFT rate (release slows down as gain increases) works well with mixed program material.

Push-button selection of the slow AGC LEVEL function (200ms attack) and/or the faster attack COMPRESSION function adapts the Co-Operator to a wide range of level control chores.

The defeatable high-frequency limiter can be used to prevent pre-emphasized material from overloading tape recorders, disk cutters, high-frequency drivers in sound systems, broadcast STLs, FM SCAs, and cassette masters which have excessive high-frequency energy. Controlling high-frequency energy permits average recording or transmission levels to be increased, yielding improved signal-to-noise ratios.

The Co-Operator also offers defeatable peak clippers which follow the high-frequency limiters. The clippers are useful when the device following the unit has an absolute "brick-wall" peak overload point (such as broadcast transmitters, digital tape, and power amplifiers). Typically, a peak clipper produces fewer audible processing-induced artifacts than a fast-attack peak limiter and is, therefore, best-suited for absolute peak protection chores.

As a result of Orban's years of research and practical experience in the broadcast signal processing field, the Co-Operator has outstandingly transparent audio performance. This performance is achieved through the use of finely-tuned control loops to eliminate the dynamic distortions that are the downfall of most conventional compressors and limiters, and through the use of a clean, Class-A proprietary VCA to ensure negligible static distortion and noise.

Easily interpreted, *independent* LED bargraphs display gain reduction and peak output levels for each channel.

Versatility Without Confusion

The Co-Operator offers unprecedented flexibility for the effective and foolproof control of levels in a wide range of professional applications. To ensure ease of use and to avoid confusing the less technical operator, set-up controls and jumpers are concealed.

The controls which calibrate the OUTPUT LEVEL meters and those which determine the actual output level are behind a pop-off door on the front panel, as are the switches which determine the pre-emphasis curves of the high-frequency limiters. The clippers, which follow the HF limiter, are strapped in or out with a jumper on the circuit board. Thus, once the Co-Operator has been set up by the installing engineer, these controls are safely out of sight. (Where the Co-Operator protects an STL, SCA, or transmitter, an optional Security Cover can prevent unauthorized adjustment of *any* control.)

Applications

Recording Studios: With the RELEASE SHAPE control set to SOFT, the outstandingly transparent Co-Operator is ideally suited for subtly reducing the dynamic range of an entire mix. The HARD setting provides effective gain riding on single tracks, increasing "punch" and intelligibility while retaining the basic feel of the performance and the apparent dynamic range of the voice or instrument. The RELEASE TIME control (a feature absent from other "invisible" compressors) is invaluable for governing the uniformity of loudness.

Digital recording and compact discs mercilessly reveal the side-effects of conventional compressors; the Co-Operator's freedom from dynamic distortion readily meets the challenge of the digital age.

Audio and Video Production: Where time is money, the Co-Operator sets up quickly and easily to protect VTRs, ATRs, or cart machines from overload during transfer. Or, use the Co-Operator on mic channels—its smoothness and silence-gating guarantee uniform, punchy voice quality without noise rush-up during pauses.

Cassette Duplication: Even with the latest advancements, such as "hot" tape and Dolby® HX Pro, there are still some masters which can cause high-frequency saturation in cassette dupes. The Co-Operator's high-frequency limiter can be set to eliminate problems caused by synthesizers, cymbals, and sibilance, while still permitting high average modulation on the cassette. The broadband AGC/leveler can be defeated to permit use of the HF limiter alone when automatic gain riding or broadband peak protection is not desired.

Sound Systems: The Co-Operator provides colorless protection for your system—whether it be a fixed installation or traveling PA. For example, in an unattended bi-amped system, the slow AGC leveler can efficiently ride gain while the faster compressor protects the system from abrupt increases in level.

Placed after the mixer and equalizer and before an active two-way crossover, the Co-Operator will protect power amps from excessive clipping and high-frequency drivers from thermal overload.

Especially with constant directivity horns, the selectable pre-emphasis of the HF Limiter can protect the horn from the boost that is required to compensate for the constant-power high-end roll-off. The Co-Operator guards against over-heating and subsequent thermal failure.

In an attended bi-amped system, the Co-Operator can be placed *after* an electronic crossover. The two channels of the unit are arranged to separately feed the power amps for LF and HF drivers. In this application the leveler would be defeated (so it would not fight with the board operator), leaving the compressor, high-frequency limiter, and peak clipper to guard the system from overload. In particularly high-SPL environments, the compressor may be uncoupled so that each channel of the Co-Operator can operate independently to deliver the maximum level to each driver without distortion.

Similarly, the Co-Operator's controlled clipping avoids clipping of the power amplifier, which can cause "sticking", instability, and other audible problems.

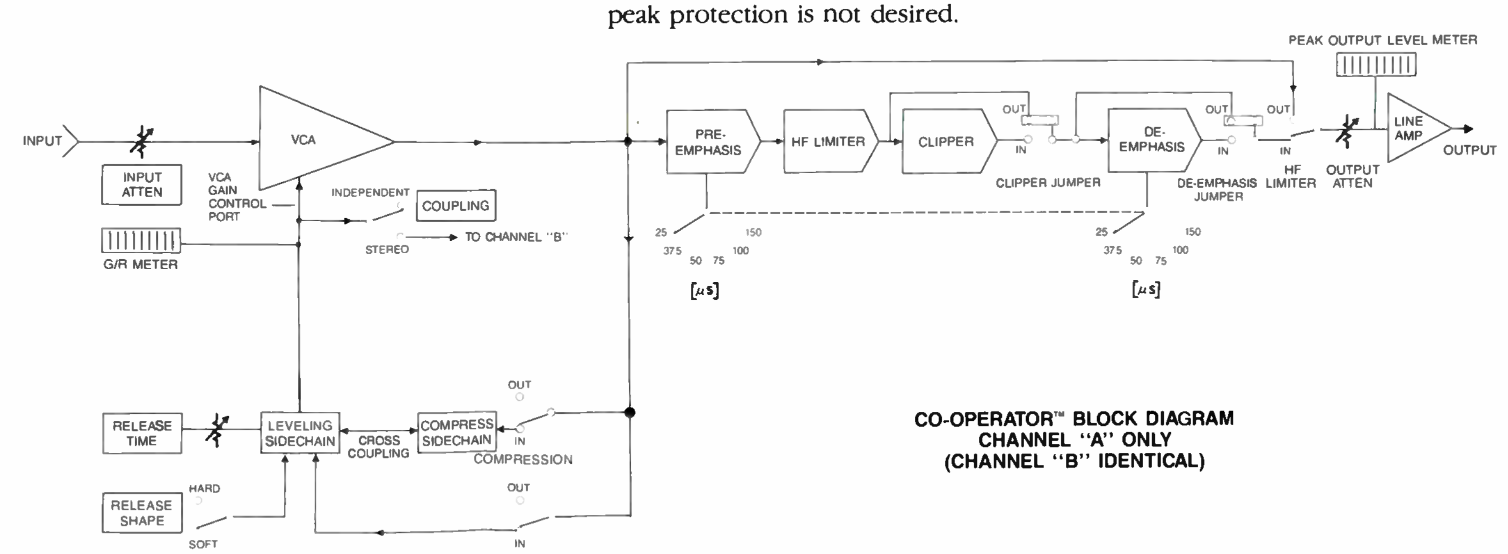
Broadcast Studio/Transmitter Link Protection: The STL is often the weak link in the broadcast chain due to barely adequate signal-to-noise ratio. Clean protective limiting ahead of the pre-emphasized STL transmitter helps maximize the signal-to-noise ratio at the receiver and prevents overload. Selectable pre-emphasis ensures matching of the high-frequency limiting function to the STL's overload characteristic.

FM Subcarriers: Voice or music SCA channels are typically pre-emphasized at 150μs. In addition to providing transparent gain riding and level control for the subcarrier, the Co-Operator can also provide the rarely found 150μs HF limiting function, which permits more efficient use of the SCA channel by eliminating overmodulation on HF peaks.

Circuit Features

The Co-Operator consists of a subtle AGC/compressor (AGC and compression can be used separately or together) cascaded with a HF limiter. To provide absolute peak control without the pumping associated with fast-attack limiter circuits, the HF limiter circuit includes a pre-emphasized clipper (which is jumper-defeatable). When the HF LIMITER switch is set to OUT, the entire HF limiter circuit (including its clipper) is removed from the signal path, ensuring maximum transparency.

Although AGC attack and release times adapt automatically to the nature of the program material to ensure the smoothest and least audible compression, the operator can fine-tune the Co-Operator to obtain the precise type of dynamic range control desired by adjusting the relative speed of the program-controlled release process. When the control of transient overshoots provided by the leveling function is insufficient, the faster "compression" function can be switched in to provide additional transient overshoot protection for material with abrupt level changes or unusually high peak-to-average ratios.



CO-OPERATOR™ BLOCK DIAGRAM
CHANNEL "A" ONLY
(CHANNEL "B" IDENTICAL)

Your Stand-Alone Stereo Encoder May Be Hurting Your Sound.

OPTIMOD-FM's built-in stereo encoder will give you much more loudness with better peak control than any other encoder.

The analog left/right outputs of OPTIMOD-FM contain a signal that is precisely and absolutely high-frequency and peak-controlled. But this carefully peak-controlled audio is not peak-controlled by the time it reaches the output of a typical encoder. A stand-alone stereo encoder includes pre-emphasis networks, low-pass filters, sometimes high-pass filters, and maybe even input transformers. These networks typically cause overshoots of 35 to 50%. On some program material they can cause overshoots of up to 6dB!

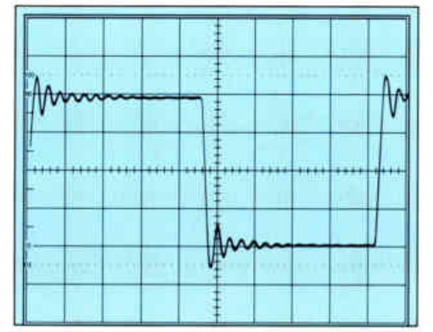


Figure 1: Typical response of uncompensated low-pass filter showing significant overshoot. (1kHz square wave)

To prevent overmodulation by these peaks, the output of the processor must be reduced by a complementary amount, which greatly reduces average loudness.

The external encoder's pre-emphasis network boosts high frequencies to either the 75µs or 50µs standard curve. In OPTIMOD-FM, the audio is pre-emphasized in the high-frequency limiter. When using an external encoder, the left/right output of OPTIMOD-FM is usually de-emphasized to provide the standard flat output. So the audio goes through three networks: pre-emphasis, de-emphasis, pre-emphasis. If they do not *precisely* match in frequency response *and* phase response—and they rarely do—the shape of the program waveform is distorted, producing overshoots.

In an external encoder, low-pass filters protect the 19kHz pilot and prevent distortion caused by aliasing-related non-linear crosstalk. Unfortunately, in most encoders, these filters are not compensated to achieve ideal group delay characteristics. So they too distort the *shape* of the program waveform as it passes through, producing overshoots —3dB is typical!

High-pass filters at 25-30Hz are sometimes included to protect the exciters' AFC. These filters add 35% overshoot! Input transformers add another 5-10% overshoot if their low-frequency response doesn't extend down to 0.3Hz or less. Most do not.

You will not see these overshoots if you measure the encoder with sine waves. But no one listens to sine waves. The overshoots will show up on square waves and program material. And they will cost your station valuable average modulation level.

OPTIMOD-FM's integrated system concept permits a different approach.

The OPTIMOD-FM system integrates the stereo encoder with the audio processor to achieve the highest average *and* peak modulation levels with the least amount of audible compression and peak limiting, compared to any other FM processor/encoder system.

The pre-emphasis networks and low-pass filters in OPTIMOD-FM are placed *before* the final peak protection circuitry. There is nothing between the peak protection circuitry and the stereo encoder to change the shape of the waveform.

The 30Hz high-pass filtering required to protect some exciters is also included.

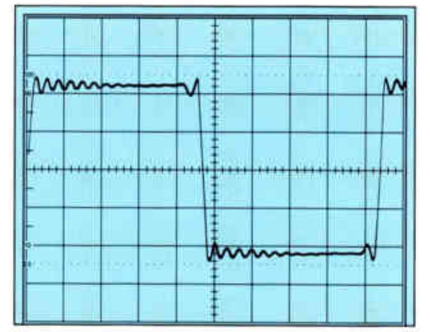


Figure 2: OPTIMOD-FM's overshoot compensated filter showing negligible overshoot. (1kHz square wave)

So by using OPTIMOD-FM's built-in encoder, instead of OPTIMOD-FM's left/right output into an external encoder, your station can be substantially louder—by 35% or more! And usually substantially brighter as well.

OPTIMOD-FM DIGITAL

8200

TECHNICAL INFORMATION

orban

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OPTIMOD-FM DIGITAL

orban

TECHNICAL INFORMATION and SPECIFICATIONS

It is impossible to characterize the listening quality of even the simplest limiter or compressor on the basis of the usual specifications, because such specifications cannot adequately describe the crucial dynamic processes that occur under program conditions. Therefore, the only way to meaningfully evaluate the sound of an audio processor is by subjective listening tests.

Certain specifications are presented here to assure the engineer that they are reasonable, to help plan the installation and to help make certain comparisons with other processing equipment.

Some of the specifications are for features that are optional or may not yet be available at the time you place your order. Consult the current price list for details.

Performance

Specifications for measurements from analog left/right input to stereo composite output and to analog left/right output are as follows:

Frequency Response (all structures, measured below gain reduction and clipping thresholds, high-pass filter off): Follows standard 50µs or 75µs pre-emphasis curve ±0.20dB, 5Hz-15kHz. Analog left/right output and digital AES/EBU output can be user-configured for flat or pre-emphasized output.

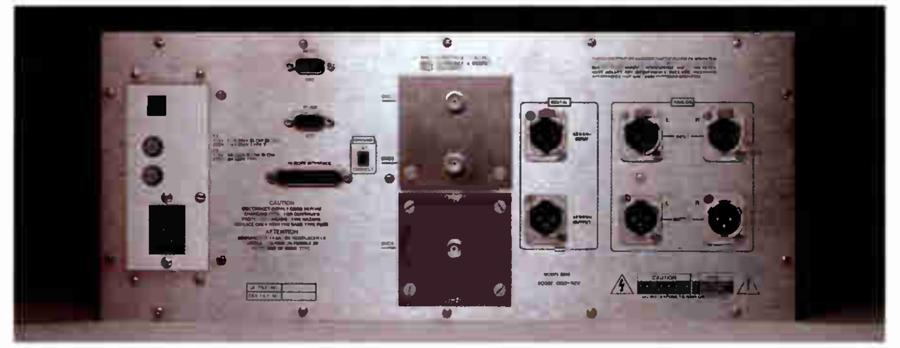
Noise: Output noise floor will depend upon how much gain the processor is set for (AGC and/or DENSITY), gating level, equalization, noise reduction, etc. It is primarily governed by the dynamic range of the A/D Converter, which has a specified overload-to-noise ratio of 97dB. The dynamic range of the digital signal processing is 144dB.

Total System Distortion (de-emphasized, 100% modulation): Less than 0.01% THD, 20Hz-15kHz. Less than 0.01% SMPTE IM Distortion.

Total System Separation: Greater than 60dB, 20Hz-15kHz; 70dB typical.

Polarity (PROTECTION or TEST structure): Absolute polarity maintained. Positive-going signal on input will result in positive-going signal on output.

A Revealing About-face...



Installation

ANALOG AUDIO INPUT

Configuration: Left and right.

Impedance: 600Ω or 10kΩ load impedance, electronically balanced, jumper selectable.

Common Mode Rejection: Greater than 70dB at 50-60Hz. Greater than 45dB 60Hz-15kHz.

Sensitivity: -30dBu to +10dBu to produce 10dB gain reduction at 1kHz.

Maximum Input Level: +21dBu.

Connector: XLR-type, female, EMI suppressed. Pin 1 Chassis, Pins 2 and 3 electronically balanced, floating and symmetrical.

ANALOG AUDIO OUTPUT

Configuration: Left and right. Flat or pre-emphasized.

Source Impedance: 30Ω, ±5%, electronically balanced and floating.

Load Impedance: 600Ω or greater, balanced or unbalanced. Termination not required.

Maximum Output Level: +24dBu into 600Ω or greater balanced load.

Connector: XLR-type, male, EMI suppressed. Pin 1 Chassis, Pins 2 and 3 electronically balanced, floating and symmetrical.

DIGITAL INPUT

Configuration: Two-channel AES/EBU-standard.

Sampling Rate: 32kHz or 48kHz, automatically selected.

Connector: XLR-type, female, EMI suppressed. Pin 1 Chassis, Pins 2 and 3 transformer balanced and floating.

DIGITAL OUTPUT

Configuration: Two-channel AES/EBU-standard.

Sampling Rate: 32kHz or 48kHz, automatically selected, following the AES/EBU digital input sampling rate.

Connector: XLR-type, male, EMI suppressed. Pin 1 Chassis, Pins 2 and 3 transformer balanced and floating.

SCA SUBCARRIER INPUT

Configuration: Subcarrier input sums into composite baseband outputs.

Input Impedance: 600Ω.

Sensitivity: 1.0Vp-p for 10% modulation of main carrier.

Connector: BNC, floating over chassis ground. EMI suppressed.

COMPOSITE BASEBAND OUTPUTS

Configuration: Two (2) outputs, each with independent OUTPUT level control, output amplifier and connector.

Source Impedance: 0Ω voltage source or 75Ω, single ended, floating over chassis ground. Jumper selectable.

Load Impedance: 37Ω or greater. Termination not required.

Level (0Ω source impedance, 75Ω or higher load impedance): Adjustable 0-8.0Vp-p with multi-turn OUTPUT LEVEL control.

Connector: BNC, floating over chassis ground. EMI suppressed.

Maximum Recommended Cable Length (0Ω source impedance): 100ft/30m RG-58A/U. Maximum permitted load capacitance 0.047µF.

DIGITAL COMPOSITE OUTPUT

Configuration: To interface directly to digital FM transmitters. Sampling rate, other interface details and connector to be determined.

PILOT REFERENCE OUTPUT

Configuration: Buffered square-wave reference for RDS or other subcarrier services.

Source: HCMOS logic level output, 0-5V peak.

Connector: On Remote Control Interface. DB-25, EMI suppressed.

REMOTE COMPUTER INTERFACE

Configuration: RS232 and RS422 interfaces to connect to IBM PC-compatible computers, directly or via modem, for remote control and metering.

Connector—RS232: DB-9, EMI suppressed.

Connector—RS422: DB-9, EMI suppressed.

REMOTE CONTROL INTERFACE

Configuration: Eight (8) inputs. User programmable to select any eight of: user presets, factory presets, stereo, mono left, mono right, reduction of modulation for SCA 1 ON, reduction of modulation for SCA 2 ON.

Voltage: 6-24VAC or DC, momentary or continuous, optically isolated. 12VDC provided to facilitate use with contact closure.

Connector: DB-25, EMI suppressed.

POWER

Requirements: Switch selectable on the rear panel, 90-130VAC or 200-250VAC, 50-60Hz; 50VA.

Connector: IEC; detachable 3-wire power cord supplied. AC is EMI suppressed.

Ground: Circuit ground is independent of chassis ground; can be isolated or connected with a rear panel switch.

Safety Standards: IEC65, UL, CSA.

DIMENSIONS

19in/48.3cm wide, 7in/17.8cm high, 15in/38.1cm deep. 4 rack units.

ENVIRONMENTAL

Operating temperature range 32-122°F/ 0-50°C Humidity 0-95% RH, non-condensing.

Setup and Operation

HELP Button: Available at all times. Push HELP, and a message will tell you what you are looking at on the screen, what can be done and how to do it.

HELP Index: Available at all times from the HELP screens. Complete listing of and access to all help messages.

SYSTEM SETUP

Function: Initial setup.

Controls: Studio chassis yes/no (defeats AGC), Pre-emphasis 50µs or 75µs, line-up (0VU or PPM reference) to clip level, I/O meters indicate reference or clip level at full scale, analog left/right input levels, analog left/right output levels, analog outputs flat or pre-emphasized, modulation compensation for STL or exciter overshoot, reduction of modulation for SCA 1 ON and reduction of modulation for SCA 2 ON.

QUICK SETUP

Function: Guided screen-by-screen setup for all required setup adjustments.

ON-SCREEN METERING

Metering can be switched to indicate gain reduction (G/R) or input/output levels (I/O).

Gain Reduction (G/R): Shows gain reduction of AGC, compressors, high-frequency limiters, and gate on/off, as appropriate to the Modular Variable Processing structure selected.

Input/Output (I/O): Left analog input (dB), right analog input (dB), left analog output (dB), right analog output (dB), composite output (% modulation). If digital I/O board is installed, metering can be switched to indicate digital I/O levels.

Pilot: 7.5-10%, in 0.5% steps.

PRESET PROGRAMMING

Function: Save processing settings for recall from the front panel, by remote control, by remote computer or by Automatic Preset Switching.

Number of User Presets: 32.

TEST PRESETS

BYPASS and G/R DEFEAT Presets: A variety of presets are available to make both US FCC-style Proof of Performance measurements and international system tests.

TONE Preset: Frequency programmable 30Hz-15kHz. Level programmable 0-133% total modulation.

BESSEL NULL TONE Preset: Frequency 13.5868kHz. Level 100% modulation. Produces 75kHz deviation on the second Bessel null.

AUTOMATIC PRESET SWITCHING

Function: Changes presets on a programmed event schedule.

Programming: Date (daily, specific day or days of the week, specific date), time, preset number.

PASSCODE SECURITY

Function: To prevent unauthorized adjustment of controls by persons without passcodes.

Number of Passcodes: 10.

Access: Each Passcode can be programmed to permit or deny access via front panel or computer to RECALL PRESET, MODIFY PROCESSING, AUTOMATION (Automatic Preset Switching) and/or SYSTEM SETUP.

REMOTE CONTROL WITH ORBAN 8200/PC REMOTE CONTROL SOFTWARE

Metering: Same as ON-SCREEN METERING, above.

Control: All user-adjustable processing parameters and Preset functions.

Circuit Characteristics

ANALOG-TO-DIGITAL CONVERTER

A/D Converter subject to change as technology improves.

Device: Crystal 5328 18-bit A/D Converter.

Performance: 97dB dynamic range (overload-to-noise ratio), per manufacturer's specifications.

DIGITAL SIGNAL PROCESSING

Device: Motorola DSP56001.

Performance: 24-bit processing. 144dB internal dynamic range.

DIGITAL INPUT CONDITIONING

Sub-sonic Fiter: Switchable in/out third-order Chebychev with 30Hz cutoff and 0.5dB pass-band ripple; -0.5dB @ 30Hz, -10.5dB @ 20Hz, -31.5dB @10Hz.

Time Dispersion Network: Switchable in/out all-pass network to make speech more symmetrical to reduce processing distortion.

DIGITAL OUTPUT CONDITIONING

Passband Response: Typically +0, -0.1dB to 15kHz.

Stopband Rejection (referenced to 100% modulation): To reduce spectrum in stereo composite baseband above 57kHz to less than -75dB.

DIGITAL-TO-ANALOG CONVERTER

Device: Analog Devices AD1864.

Performance: 18-bit. 4X over-sampled output. Linear-phase reconstruction filters.

PILOT

Frequency: 19kHz.

Stability: ±0.005%. Contact the factory if greater stability is required for special applications.

Injection: Adjustable, 7.5-10%.

STEREO BASEBAND GENERATOR

Noise (de-emphasized, referenced to 100% modulation): Less than -100dB, 20Hz-15kHz.

Distortion (de-emphasized, 100% modulation): Less than 0.005% THD, 20Hz-15kHz; less than 0.005% SMPTE Intermodulation Distortion.

Stereo Separation: Greater than 60dB, 20Hz-15kHz; 70dB typical.

Crosstalk—Linear (referenced to 100% modulation): Less than -75dB, main channel to sub-channel or sub-channel to main channel.

Crosstalk—Non-linear (referenced to 100% modulation): Less than -80dB, main channel to sub-channel or sub-channel to main channel.

38kHz Sub-carrier Suppression (referenced to 100% modulation): Greater than 70dB; 75dB typical.

76kHz and Sideband Suppression (referenced to 100% modulation): Greater than 70dB.

CIT-25 Composite Isolation Transformer

Ground loops are a problem in some transmitter plants. A ground loop among OPTIMOD-FM and one or more exciters may occur, causing an increase in hum and noise. This is especially likely when OPTIMOD-FM is installed some distance from the exciter.

The Composite Isolation Transformer provides the solution. Designed to be installed adjacent to each exciter, it provides ground loop isolation between the OPTIMOD-FM composite output and the exciter, and presents OPTIMOD-FM with a balanced floating load.

Performance

Frequency Response: +0.01, -0.03dB, 30Hz-53kHz.

Group Delay: Deviation from linear phase less than ±0.3°, 30Hz-53kHz.

Separation: Greater than 50dB, 30Hz-10kHz, greater than 45dB, 10kHz-15kHz.

Gain: Adjustable from full attenuation to 0dB.

Installation

LOCATION

As close to FM exciter as practical.

INTERCONNECT BETWEEN OPTIMOD-FM AND TRANSFORMER

Interface at OPTIMOD-FM: Adapter cable, BNC male to 3-pin XLR-type male cable connector supplied.

Cable: Two-conductor foil shielded audio cable, Belden 8451 or equivalent. Maximum length 50ft/15m. 3-pin XLR-type male and female cable connectors supplied.

COMPOSITE BASEBAND INPUT

Connector: 3-pin XLR-type female. Pin 1 capacitively coupled to chassis, Pins 2 and 3 transformer balanced, floating and symmetrical.

Maximum Level: 4.4Vp-p.

COMPOSITE BASEBAND OUTPUT

Connector: BNC, shell insulated from chassis.

INTERCONNECT BETWEEN TRANSFORMER AND EXCITER

Maximum Recommended Cable Length: 6ft/1.8m RG-58A/U cable or similar, to avoid excessive RF pick-up on cable. Cable supplied.

Exciter Input: 1kΩ or greater; 1000pF or less. Replace 50Ω or 75Ω termination resistor, if present, with 1kΩ resistor, supplied.

DIMENSIONS

7in/17.8cm wide, 3in/7.6cm high, 1.72in/4.4cm deep.

Warranty

AKG Acoustics, Inc. warrants Orban products against defects in material and workmanship for a period of one year from the date of original purchase for use, and agrees to repair or, at our option, replace any defective item without charge for either parts or labor.

This warranty is valid for the original purchaser only.

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 inadequate repairs performed by unauthorized service centers.

Subject to limitations stated in our United States Limited Warranty and Standard International Warranty.

Orban engineers are constantly working to improve the quality of our products. Specifications are, therefore, subject to change without notice.

orban®

Radio Broadcast Products
USA Suggested Introductory List Prices

Effective April 15, 1991

OPTIMOD®-FM 8200

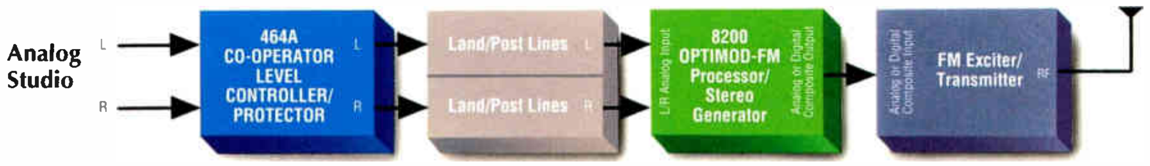
		USA INTRODUCTORY LIST PRICE
8200/U2S	OPTIMOD-FM 8200 DIGITAL with two DSP cards, stereo coder, Protection Structure, Two-band Purist Structure, Two-band Processing Structure, 120V, switchable 50µ or 75µ.	$7,400.00
8200/E2S	OPTIMOD-FM 8200 DIGITAL with two DSP cards, stereo coder, Protection Structure, Two-band Purist Structure, Two-band Processing Structure, 230V, switchable 50µ or 75µ.	$7,400.00
8200/U3S	OPTIMOD-FM 8200 DIGITAL with three DSP cards, stereo coder, Protection Structure, Two-band Purist Structure, Two-band Processing Structure, Multi-band Structure, 120V, switchable 50µ or 75µ.	$9,820.00
8200/E3S	OPTIMOD-FM 8200 DIGITAL with three DSP cards, stereo coder, Protection Structure, Two-band Purist Structure, Two-band Processing Structure, Multi-band Structure, 230V, switchable 50µ or 75µ.	$9,820.00

Orban, a division of AKG Acoustics, Inc.
1525 Alvarado Street, San Leandro, CA 94577
Tel 415/351-3500 Fax 415/351-0500

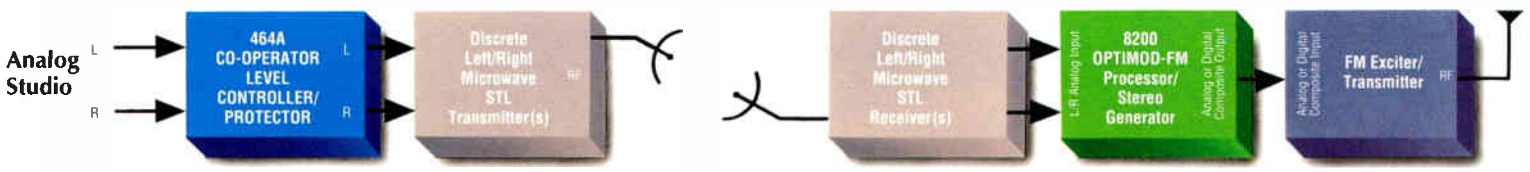
Common System Configurations.

Co-located analog studio and analog or digital transmitter

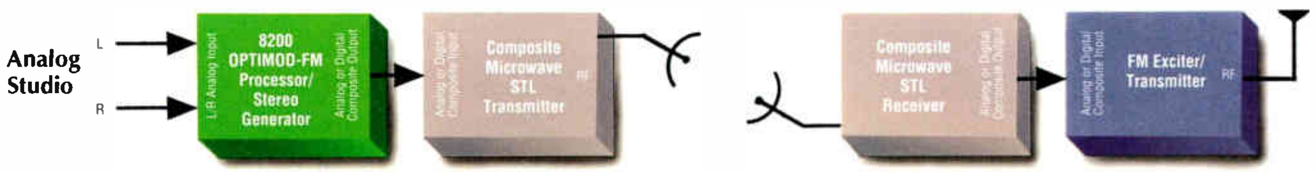
Co-located digital studio and analog or digital transmitter



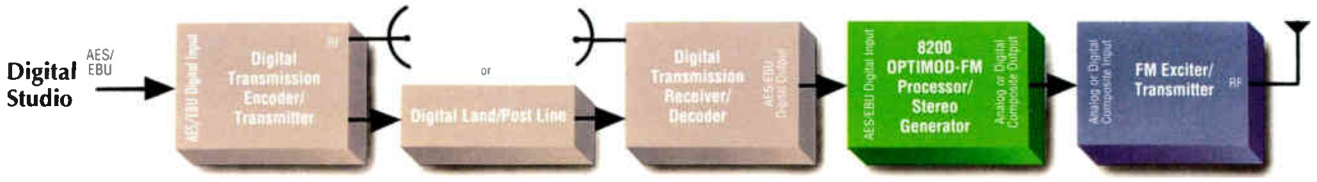
Program feed from analog studio to analog or digital transmitter via discrete analog land/post lines (or digital lines with analog left/right input/output)



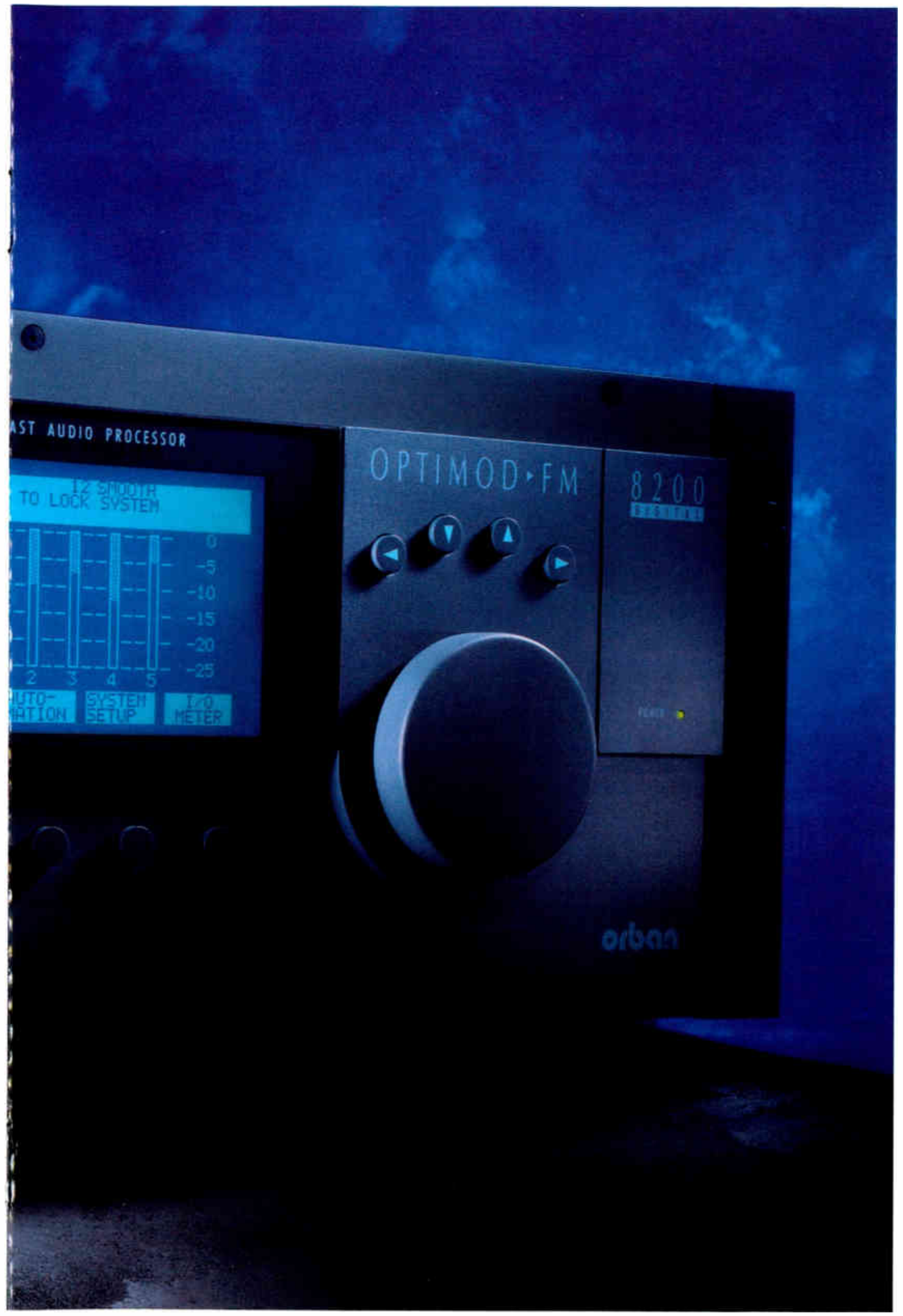
Program feed from analog studio to analog or digital transmitter via left/right analog STL (or digital STL with analog left/right input/output)



Program feed from analog studio to analog or digital transmitter via composite analog or digital STL



Program feed from digital studio to analog or digital transmitter via digital microwave or land/post line



Bob Orban and his engineering staff worked tirelessly for more than half a decade to design a true successor to the worldwide, industry-standard OPTIMOD-FM 8100A. They refused to settle for less than another breakthrough in audio quality, performance, control and flexibility.

The result. The OPTIMOD-FM 8200 Digital Audio Processor.

The OPTIMOD-FM 8200 Digital Audio Processor provides complete audio processing and transmitter protection for FM broadcast. The 8200 interfaces with all commonly found transmitters and studio-to-transmitter links, and is designed to grow with future advances in digital FM technology.

In the tradition of our analog OPTIMOD-FM 8100A, OPTIMOD-FM 8200 provides a strong, high-quality sound that attracts and holds listeners. Its fully-digital audio processing improves the quality and clarity of the successful "OPTIMOD sound," while adding changeable processing structures, programmability, expandability and a PC interface.

OPTIMOD-FM 8200.
The first FM audio processor to combine the **power** of OPTIMOD with the **power** of digital.

OPTIMOD·FM
DIGITAL

The Power of OPTIMOD.

Creating a "sound" that attracts and holds the largest possible audience is the bottom line in the radio business.

Talk to general managers, program directors and chief engineers at any of the thousands of FM stations around the world that have OPTIMOD-FM on the air. They'll tell you that OPTIMOD-FM allows them to create a strong, solid, high-quality signal that attracts and holds listeners. That's why OPTIMOD-FM is the most widely used FM stereo processor in the world.

OPTIMOD-FM's sound is *open and natural* with a uniquely favorable combination of impact, clarity, loudness and brightness. To further fine-tune their sound to their target audience and market position, some stations have added OPTIMOD-FM's companion Six-Band Limiter. It offers them the ability to broadcast a *louder* sound with *fewer* processing artifacts.

The OPTIMOD-FM 8200 Digital Audio Processor takes the OPTIMOD standard to an even higher level.

Sutro Tower, San Francisco, California. At 977 feet, Sutro Tower is the highest structure in San Francisco.

The Unique, Successful OPTIMOD Sound Gets Better.

The 8200 carries on the tradition of louder, cleaner and brighter audio. Our digital signal processing improves the OPTIMOD sound with high-frequency and peak controls that are not achievable in analog circuitry. Two years of 8200 development time were spent on achieving the most effective peak control on the market.

The now classic OPTIMOD-FM 8100A was designed with a single guiding principle—to optimize loudness and create a natural, musical sound that attracts and holds an audience.

OPTIMOD-FM—Not an Ordinary Compressor/Limiter.

The OPTIMOD-FM system integrates the stereo encoder with the audio processor to achieve the highest average and peak modulation levels with the least amount of audible compression and peak limiting, compared to any other FM processor/encoder system.

Orban's several patented distortion-cancelling clipping systems control peaks to prevent over-modulation, without the "peak limiter sound." Peaks are controlled precisely, with no audible distortion, and no affect on the average level.

K101, San Francisco

as important, however, is the room it allows for future growth. Room for growth means more than additional expansion slots that will accommodate more DSP chips or cards. It means the processor's fundamental ability to absorb and adapt to major technological innovations.

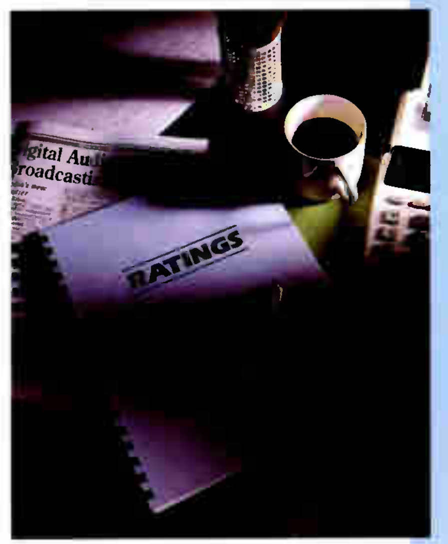
OPTIMOD-FM 8200 is such a machine. You can put it to maximum use today, but it also provides a platform for future breakthroughs in digital signal processing technology. It is an investment in the digital future.

Bottom Line Impact.

OPTIMOD-FM 8200 will have a direct, positive effect on your station's bottom line because:

- It will increase audience size and share with a strong, clear signal that will maximize modulation without risk of over-deviation.
- It will create an open and natural sound, ideally processed for your format or daypart, that will attract more listeners for longer periods of time.
- Its flexibility, programmability and ease of adjustment will cause it to be put to maximum use.

The secret to winning the ratings game is to create a sound that reaches the largest possible area and attracts and holds your desired audience at all times of the day. OPTIMOD-FM 8200 gives station management a powerful competitive weapon that easily and automatically tailors a station's sound to maximize market share.



- It will keep your capital equipment costs down with flexible digital technology that allows you to upgrade and expand.

No other digital signal processor in the world improves upon the sound of the original OPTIMOD-FM or delivers on the promise of digital control and ease-of-use. The 8200 does both.

Don't Be the Last in Your Market...

to take advantage of the power, potential and profitability of OPTIMOD-FM 8200. A personal, hands-on demonstration of the 8200 will give you the opportunity to hear, see and touch the future of audio processing.

OPTIMOD-FM 8200. A technological breakthrough with bottom line impact.

AKG stands for "Akustische U. Kino-Geräte." Established in 1947, this Austrian firm was started by two engineers and grew into a multinational company with 103 national representatives and distributors in nearly all countries of the world.

AKG is a major innovator, holding more than 1400 patents worldwide, including over 300 in fundamental areas of audio transduction and processing.

In April 1989, Orban, long known as the industry leader in broadcast audio processors, became an operating division of AKG Acoustics, Inc.

AKG®

The Digital Future Starts Here.

OPTIMOD-FM 8200 is fundamentally better than any other broadcast audio processor. Given the success of the analog OPTIMOD-FM 8100A, that should come as no surprise. The 8200 is to digital broadcast processors what the original OPTIMOD was to every other processor on the market.

A quantum leap ahead.

The Bigger the Market, the Bigger the Bottom Line.

In the competitive FM market, stations live and die by the ratings. Ratings mean income, and the better the ratings, the better the income. That's a fact of life in the business. And there are three elements to achieving and maintaining high ratings: **programming**, fine-tuned to your target audience and market position; **reach**, to cover the widest geographic area; and **retention**, to hold an audience and get them to tune in again and again.

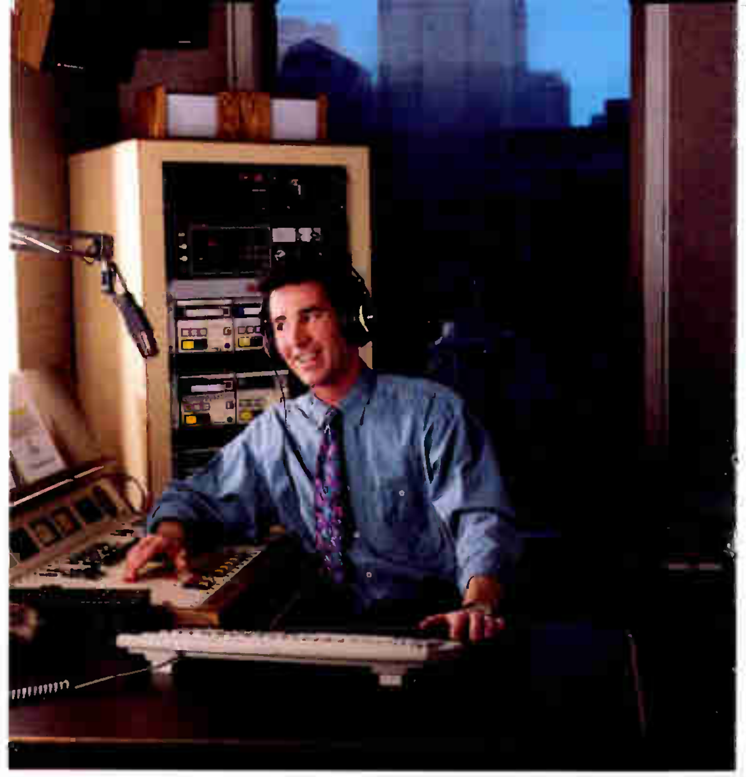
OPTIMOD-FM 8200 gives station management a powerful competitive weapon—to reach and retain the largest possible audience.

The 8200's fully-digital audio processing improves the quality, clarity and penetration of the successful "OPTIMOD sound" while adding changeable processing structures, programmability, expandability and a PC interface.

What is the value of true digital processing and control? In addition to improving the quality of the processed signal, digital makes the OPTIMOD-FM more user-friendly, more programmable, more flexible. Simply put, because OPTIMOD-FM is easier to adapt to a station's programming needs, it will produce more benefit, more of the time, than was previously possible.

Limitless Possibilities.

The most important consideration in selecting an audio processor is what it can do for your station today. Nearly

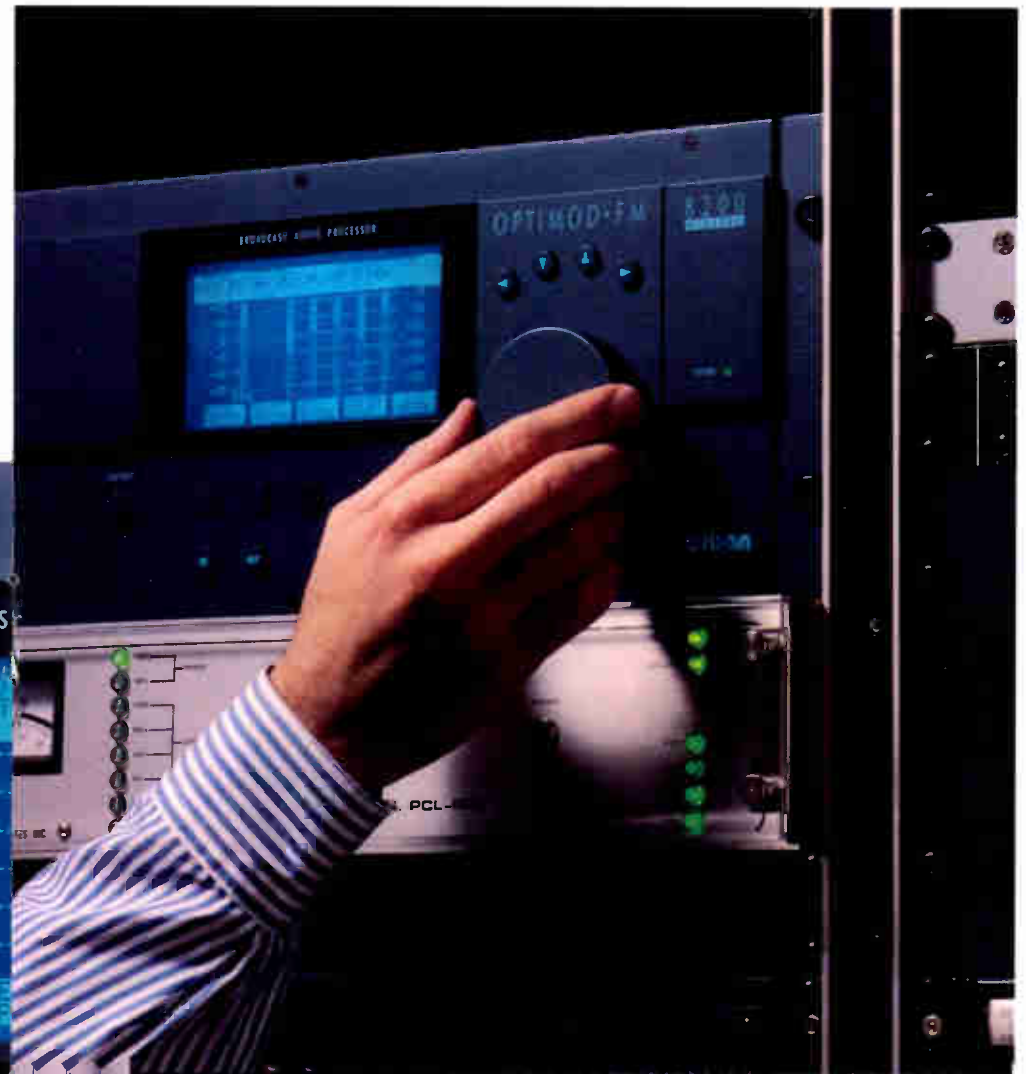


With OPTIMOD-FM 8200, your listeners will enjoy a clear, bright, appealing sound that is ideally processed for the format and easy to listen to for long periods of time.

The 8200's digital processing power makes possible a spectrum of system flexibility, programmability, control and ease-of-use features never before available in any audio processor.

OPTIMOD-FM's high-frequency control is almost never audibly apparent. The processing and the high-frequency control operate as a system to provide a clean, open sound with much more subjective brightness.

By integrating the stereo encoder with the audio processing, OPTIMOD-FM eliminates the overshoot problems that waste valuable modulation in traditional external encoders.



OPTIMOD-FM 8200 is a completely digital audio processing system for FM broadcast. The 8200 fulfills all of a station's processing needs: automatic gain control, compression, peak modulation control and stereo encoding.

Digital Signal Processing (DSP) cards and Program Memory Modules (PMM) make the 8200 fully expandable. By installing new boards and modules, future software upgrades and processing structures can be added easily—without purchasing a new box!

Built for Reliability and Durability.

OPTIMOD-FM 8200's use of digital integrated circuits increases the unit's reliability and ease of service. Proven, modular components and stringent reliability testing deliver maximum system uptime. During manufacture, DSP cards and finished systems pass rigorous burn-in and heat cycling tests.

Backed by AKG's worldwide service support centers.

The 8200's modular construction makes servicing the unit easy. All circuitry, even the power supply, is on plug-in boards or modules for easy troubleshooting and maintenance. Replacement cards and modules are available from AKG service centers worldwide.

The Power of Digital.

The power of digital propels the 8200 to new levels of performance and functionality. OPTIMOD-FM 8200 is a true digital audio processor—the audio is digitized and all control functions are digital.

Digital audio processing provides the tools for a dramatic improvement in sound quality, and the capability of instant switching between radically different processing structures.

Digital control makes possible a spectrum of system flexibility, programmability and control previously unavailable at any price.

Modular Variable Processing (MVP) Structures.

MVP structures, a flexible processing concept pioneered by Orban and achievable only in digital processing, allow the 8200 to change its sound with the push of a button.

Each MVP structure is the software equivalent of a dedicated processor. In a typical 8200, one MVP acts as a two-band processor, another as a multi-band processor, and a third MVP functions as a transparent protection limiter.

Easy to Adjust.

Versatile processors have many controls. OPTIMOD-FM is no exception. This degree of flexibility is required by some stations to get precisely the sound that they want and need for their competitive market.

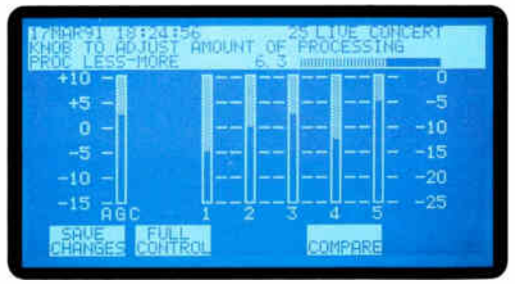
RECALL any one of the factory installed or station installed presets. They can be recalled from the front panel or from a local or remote computer. Or they can be automatically recalled on a programmed schedule.

Many stations, however, just set the controls to one of the factory's recommended settings. So with the 8200, the factory recommended settings are built-in. Each MVP structure can be easily adjusted with just one control, LESS<>MORE, to access hundreds of factory recommended settings. Using LESS<>MORE precisely tunes all of the processing parameters together to produce more or less processing, with all parameters ideally set.

Easy to Customize.

Create a custom sound by using FULL CONTROL to fine-tune each parameter individually. Then store the adjustments for this sound as a Station Installed Preset.

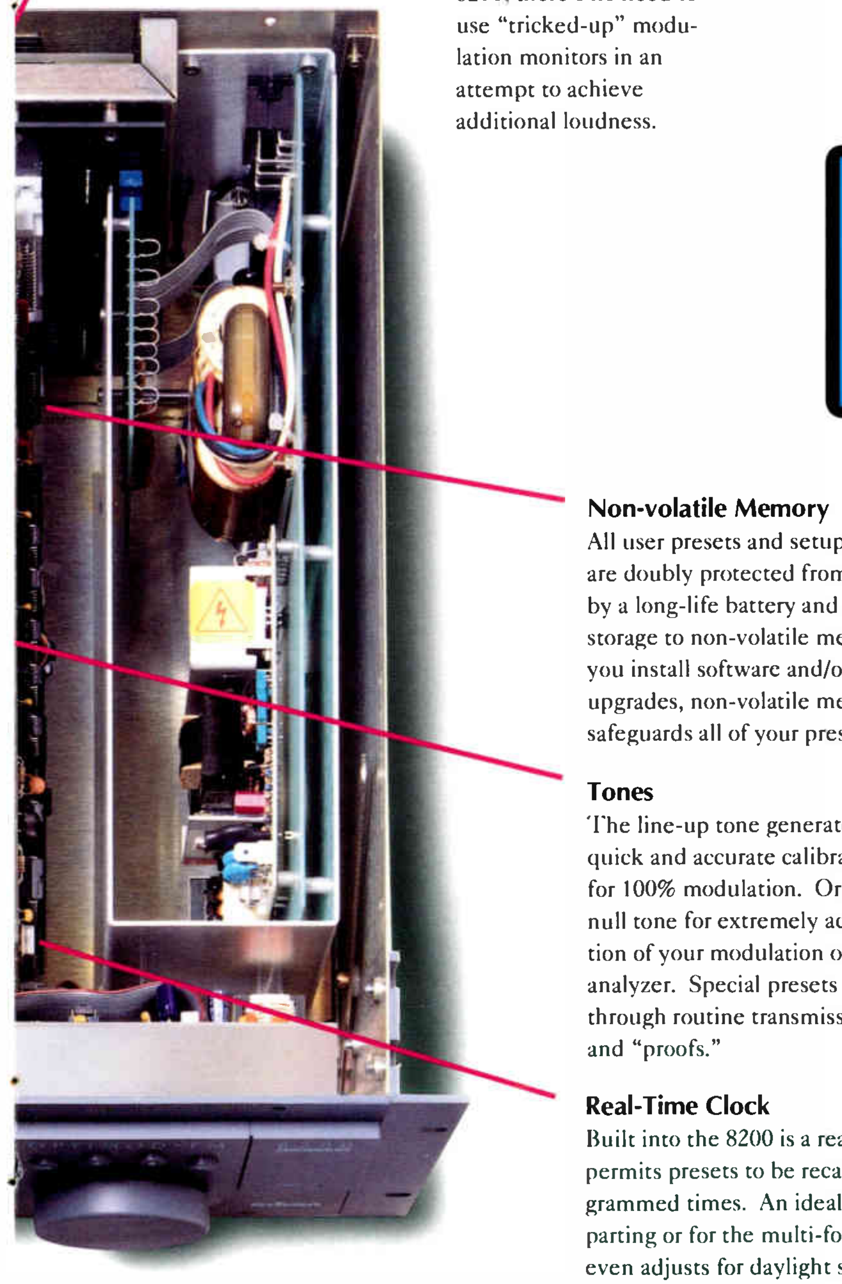
LESS<>MORE: Adjust all audio parameters with one control. Turn toward LESS for a smooth, easy, less-processed sound. Turn toward MORE for a louder, punchier, more-processed sound. LESS<>MORE is like a hundred factory settings. Getting the sound you want is as easy as turning one knob.



FULL CONTROL: For the engineer who wants to customize, FULL CONTROL allows access to every user-adjustable parameter in each MVP structure.

Outstanding Peak Control to Prevent Over-modulation

Peak limiting is inherently more difficult to do in digital than in analog. It took Orban's experienced DSP engineers more than two years to develop the algorithms that allow the 8200 to achieve its extremely tight peak control. With the 8200, there's no need to use "tricked-up" modulation monitors in an attempt to achieve additional loudness.



Non-volatile Memory

All user presets and setup information are doubly protected from memory loss by a long-life battery and by simultaneous storage to non-volatile memory. When you install software and/or hardware upgrades, non-volatile memory also safeguards all of your presets.

Tones

The line-up tone generator facilitates quick and accurate calibration of the 8200 for 100% modulation. Or use the Bessel-null tone for extremely accurate calibration of your modulation on a spectrum analyzer. Special presets breeze you through routine transmission system tests and "proofs."

Real-Time Clock

Built into the 8200 is a real-time clock that permits presets to be recalled at programmed times. An ideal feature for dayparting or for the multi-format station. It even adjusts for daylight savings time!

Expandability

The 8200 is expandable. Add more DSP processing power, as needed, to accommodate future software upgrades and new processing structures.

Automatic Preset Switching:

Recall processing presets on a programmed event schedule. This function is ideal for daypart processing or for multi-format stations.

QUICK SETUP is a guided screen-by-screen walk-through for all required setup adjustments.

Take a Tour of the OPTIMOD-FM 8200.

Serviceability
The 8200 is fully modular. All electronics, including the power supply, are contained on plug-in cards or modules for ease of service or exchange.

Versatile Remote Control Interface
Use your existing transmitter remote control to activate one of eight pre-assigned processing presets. Or connect an IBM-compatible computer locally, or from your studio via modem.

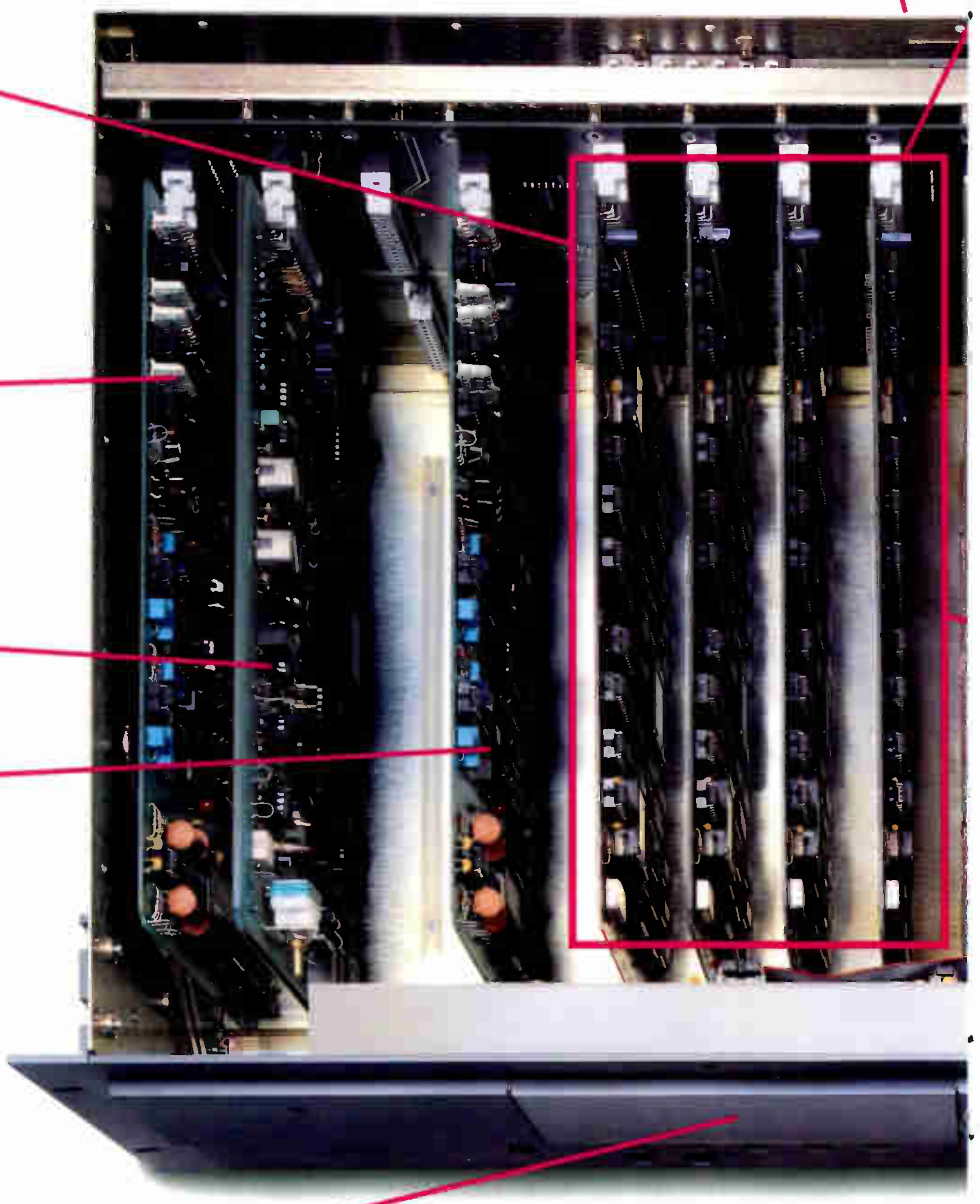
DSP Algorithms
Full Digital Signal Processing gave Orban's engineers the opportunity to take a fresh, new approach to audio processing technology. They started with the signal processing qualities that made our analog OPTIMOD-FM so successful, improved upon them using digital design, then added new features and functions not possible in analog.

Stereo Encoder
Orban's Digital Hadamard Transform Baseband Encoder™ produces an extremely well-controlled spectrum, with vanishingly low noise and distortion and outstanding separation.

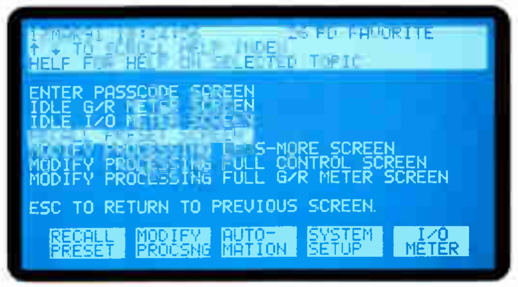
Analog Input/Output
Stereo analog balanced inputs and outputs are standard.

Digital Input/Output
An optional Digital Input/Output Card provides stereo AES/EBU input and output at the standard sampling rates of 32kHz and 48kHz.

Security Protection
The 8200's controls can be locked out, requiring a passcode for access. Passcodes can be set for varying levels of access—from RECALL PRESET only to full setup and programming.



LCD Screen
A large Liquid Crystal Display (LCD) makes setup, adjustment and programming of the 8200 easy. The screen clearly shows all metering functions of the Modular Variable Processing structure in use. Only those controls related to what you are doing are shown.



HELP is always available with the push of a button. Push HELP and a message will tell you what you are looking at on the screen, what can be done and how to do it. Or scan the Help Index that will direct you to instructions for any operation. Everything you need is on the screen.

Save up to 32 PRESETS to recall at any time. Recall a PRESET from the front panel, by remote contact closure, or by computer interface.

Automatic Preset Switching.

This feature changes the 8200's processing on a programmed schedule. It is ideal for dayparting—changing the processing to meet the requirements of drive time, mid-day at the office, or serious evening listening. Automatic Preset Switching allows stations that broadcast different formats to optimize the processing to the programming throughout the day.

User-Friendly Interface.

A large Liquid Crystal Display (LCD) makes setup, adjustment and programming of the 8200 easy. The screen clearly shows all metering functions of the Modular Variable Processing structure in use.

Push one of the clearly labeled softkeys to RECALL a preset, to MODIFY processing, to program Automatic Preset Switching, or to access system SETUP.

HELP is always available at the push of a button. If you get lost, ESC will always bring you back home. There is no need to find a complex instruction manual to adjust and program the 8200. Step-by-step instructions are on the screen.

Upgrade and Expand.

MVP structures and the control system program are stored on a plug-in module, making the 8200 easily upgradable.

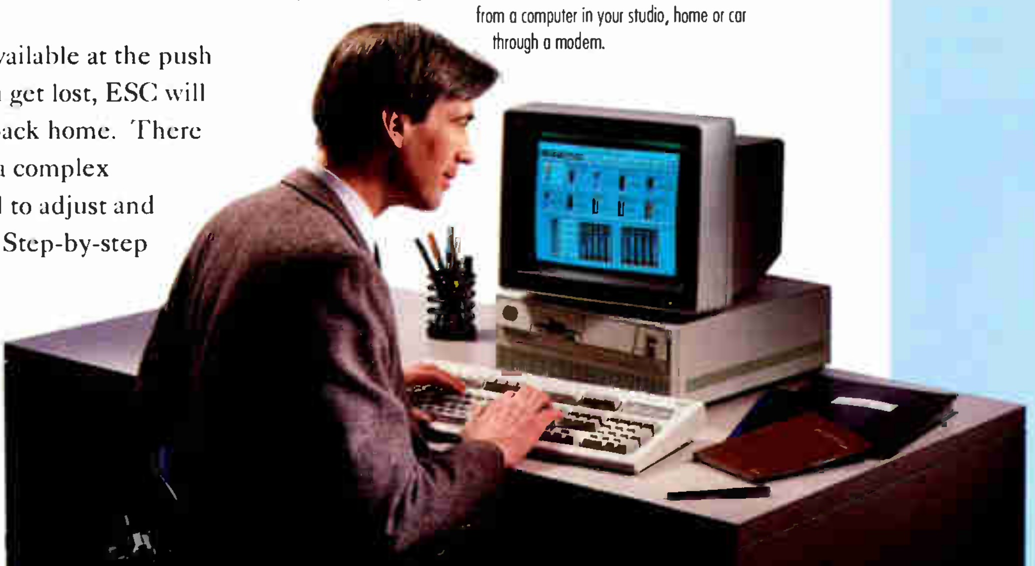
DSP cards can be added as needed when future software upgrades and additional processing structures require more processing power.

Since all processing is accomplished through software, the 8200 will always be upgradable. Change your sound by changing the software, not the audio processor!

Versatile Control.

OPTIMOD-FM 8200 can be easily controlled from the front panel. Connect it to your current transmitter remote control and select up to eight presets. Or connect it to an IBM-compatible PC, using optional software, to get full control with on-screen displays. With a standard computer modem, the 8200 can be controlled from a computer in your studio, your home, or even your car.

Using optional software, OPTIMOD-FM 8200 can be fully operated from an IBM-compatible PC. Anything that can be done from the 8200's front panel can be done from a computer in your studio, home or car through a modem.



The Motorola DSP56001 is a special purpose, programmable digital signal processing chip that handles the most difficult audio calculations at the lightning speed of 13.5 Million Instructions Per Second (MIPS). Six to twelve DSP56001 digital signal processors are used in the 8200, depending upon the configuration.

Modular Variable Processing (MVP) Structures.

With conventional analog processing, multiple structures require multiple boxes. With the MVP architecture, processing structures can be changed with the push of a button! OPTIMOD-FM 8200 can be...

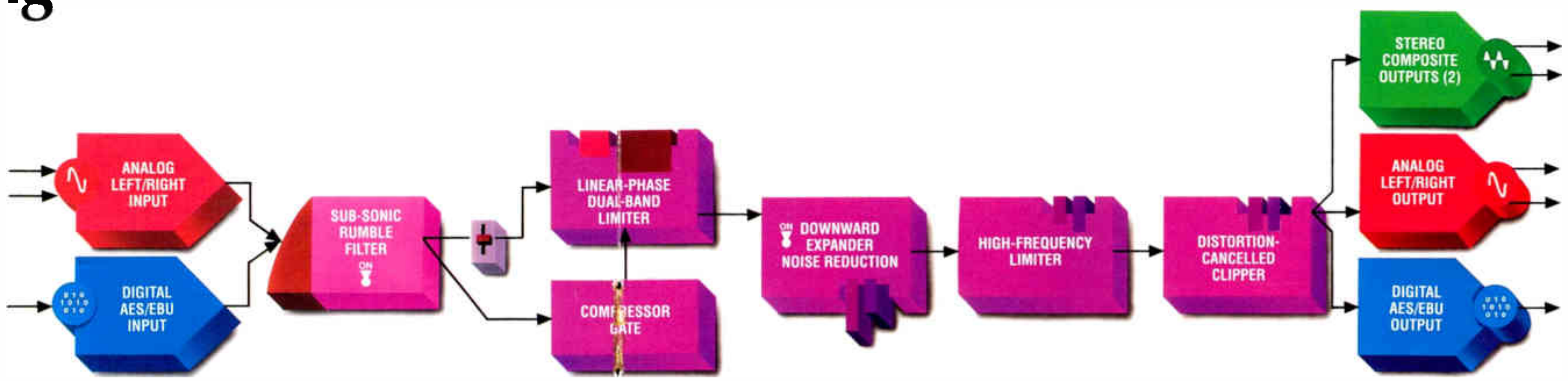
...a protection processor, for extremely effective modulation control that is totally inaudible;

...a two-band processor, for adult-oriented formats requiring a consistent, yet transparent sound;

...a multi-band processor for major market "competitive" processing.

An OPTIMOD 8200 can be purchased with any or all of our currently available MVP structures. MVP pre-processors are now under development for delivery soon.

PROTECTION MVP

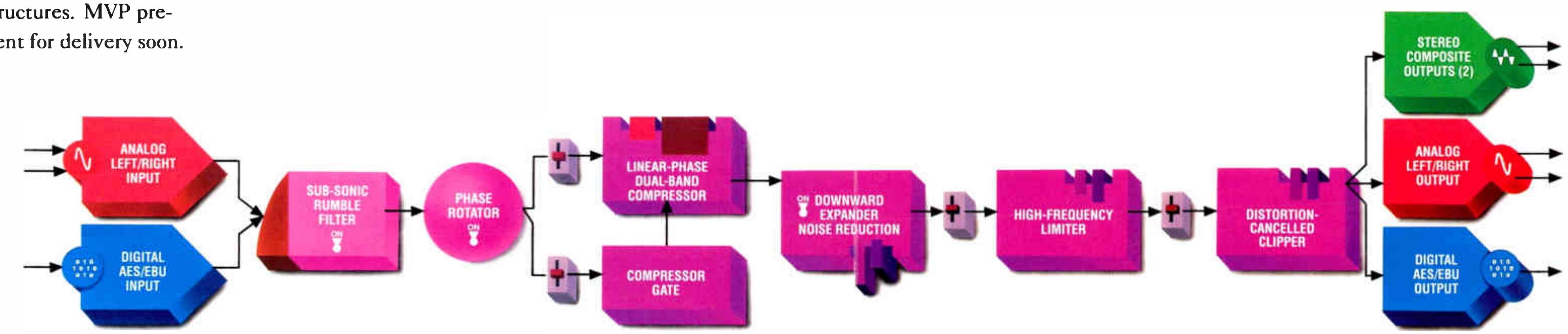


The Protection MVP is similar in operation to the Orban 4000A Transmission Limiter. It is ideal for broadcasting classical music with its full dynamic range, or for any other programming where absolute transparency of processing is desired.

Operate the Protection MVP below the threshold of limiting for all but the crescendos or for reducing dynamic range, and the sound on the air will be virtually indistinguishable from the original.

Of course, the Protection MVP is not true "processing." It will not make the program loud—it prevents overmodulation, and does so without the nasty "artifacts" typical of conventional peak limiters.

TWO-BAND MVP

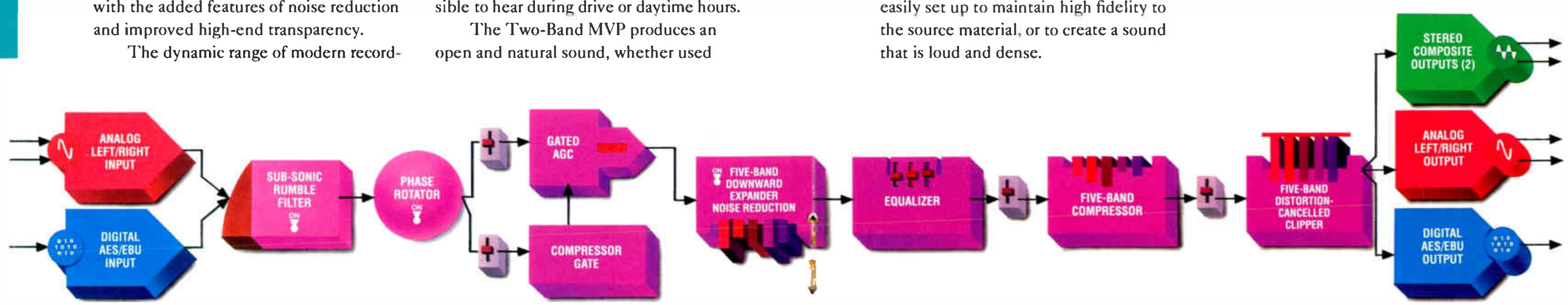


The Two-Band MVP is similar to the Orban 8100A processing that has helped make thousands of stations successful—with the added features of noise reduction and improved high-end transparency.

The dynamic range of modern recordings can be so wide that when listening levels are adjusted for comfortable loudness on loud passages, softer passages are impossible to hear during drive or daytime hours.

The Two-Band MVP produces an open and natural sound, whether used merely for light control over levels, or for the heavier processing often desired for some popular music formats. It can be easily set up to maintain high fidelity to the source material, or to create a sound that is loud and dense.

MULTI-BAND MVP



The new Multi-Band MVP is the ultimate processing for the competitive major market sound. Designed by Bob Orban and Greg Ogonowski, it gives your station more punch, more consistency, more presence, and more brightness, without pumping or voice distortion.

The Multi-Band MVP lets you set the speed limit. FAST creates a synthetic sound, an illusion, a sound that is distinctly different. A sound with lots of punchy dynamic bass, lots of up-front presence, lots of unrestricted transparent bright highs. Vocals stand out. It is the ultimate sound for Contemporary Hits Radio (CHR).

SLOW creates a very open sound with slow adjustment of frequency balance for consistency. There is a distinct improvement in clarity—the sound is very life-like. SLOW produces the effect of a wide dynamic range. But make no mistake, SLOW is a loud sound, competitive in any market. Ideal for adult contemporary, beautiful music and talk formats.

With Multi-Band MVP you can equalize the frequency balance, and control the amount of wideband AGC, multi-band compression (density), clipping and noise reduction.

Upgrade and Expand.

MVP structures and the control system program are stored in a plug-in module, which makes the 8200 easily expandable. Add DSP cards as needed when future software upgrades and additional MVP structures require more processing power.

Since all processing is accomplished through software, the 8200 will always be upgradable. Change your sound by changing the software, not the audio processor.

OPTIMOD-TV® 8182A

MONO OR STEREO

Function of OPTIMOD-TV:

Model 8182A is Orban's second-generation TV audio processor: It is an integrated signal-processing *system* which replaces conventional compressors, limiters, and clippers to precisely control audio modulation without introducing audible artifacts. Based on the popular OPTIMOD-FM Model 8100A, the 8182A offers the TV broadcaster the same superb quality that has made the 8100A so popular among FM radio broadcasters, plus Loudness Control and Hilbert-Transform clipping—features which tailor it perfectly to the unique requirements of television audio.

The 8182A is also ideally suited for conditioning signals prior to satellite uplinks, as well as for audio processing in other specialized communications applications where extremely high audio quality is desired, the audio spectrum must be limited to a 15kHz bandwidth, and the channel peak overload point is abrupt and must not be exceeded.

This brochure provides a technical description. However, it can't adequately describe the most important feature of the 8182A: its natural *sound*, and its ability to handle typical television audio feeds—from master tape to 16mm optical film—smoothly and gracefully without introducing processing artifacts. These days, consumers are accustomed to good sound, and OPTIMOD-TV helps you provide audio quality which augments the video quality you have worked so hard to achieve.

Briefly, the OPTIMOD-TV system performs these functions:

☐ It rides gain over a range of as much as 25dB, compressing dynamic range and compensating for gain-riding errors. Gain riding and compression are virtually undetectable because of advanced program-controlled time constants, level-dependent gating, and multi-band compression.

☐ It controls excessive perceived loudness by means of a complex loudness estimating circuit (which can be enabled and defeated by remote control). This circuit, licensed from CBS Technology Center, incorporates the results of their second major loudness research project (1978-1980). On-air tests of the controller have resulted in a substantial reduction or elimination of viewer complaints regarding excessively loud commercials.

☐ It controls potential interference to video and/or future stereo services by means of bandwidth-limiting 15kHz lowpass filters incorporating full overshoot compensation. OPTIMOD-TV thus provides extremely tight control over peak modulation, preventing overmodulation and controlling its output spectrum simultaneously.

☐ The OPTIMOD-TV compressor is a dual-band design which can be operated with the bands independent of each other ("independent"), or such that the bands are coupled and ordinarily track each other ("wideband"). When operated in "independent" mode, OPTIMOD-TV makes audio quality more consistent by correcting frequency balances between bass and midrange material. When operated in "wideband" mode, it will preserve frequency balances and will produce an output which sounds like its input.

☐ It prevents peak overload and overmodulation due to the effects of the preemphasis curve.

An accessory port is included to interface the noise reduction encoder required for TV stereo. In addition, an external TV stereo generator will be needed.

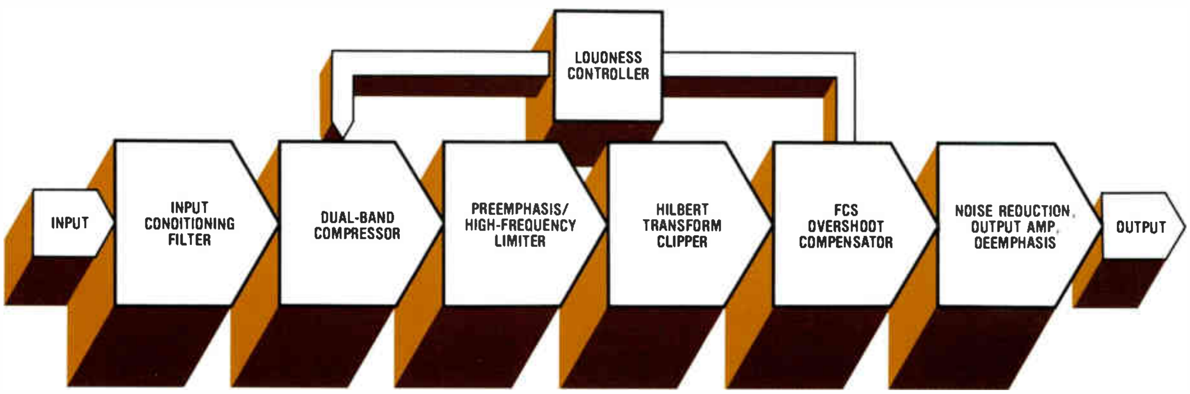
Internal jumpers determine if the active-balanced $\pm$10dBm outputs are to be flat or preemphasized and whether they are to be in conventional left/right or in sum-and-difference form.

Ordinarily OPTIMOD-TV should be fed unprocessed audio. Additional compression and/or other audio processing would not be desirable except as might be applied to individual microphone channels in a live production environment or to other sources requiring special processing.

Split Configuration: An alternate, dual-chassis configuration permits the Dual-Band Compressor to be operated separately from the remainder of the circuitry. The Dual-Band Compressor can be placed at the studio side of the STL (telephone line, dual microwave, or FM subcarrier on a video STL) to protect the STL from overmodulation and to put most operating controls at the studio.

This configuration consists of a basic chassis in conjunction with the Model 8182A/ST Accessory Chassis. The 8182A/ST accepts several cards from the main chassis which are replaced by jumper cards.

However we discourage use of the split configuration if the gain of the STL cannot be maintained $\pm$0.75dB. In this case we recommend an Orban Compressor/Limiter (such as the 424A) at the studio side of the STL to protect against STL overload.



SYSTEM BLOCK DIAGRAM

(ONE CHANNEL DEPICTED)

3. Preemphasis And High Frequency Limiter: The output of the Dual-Band Compressor is applied to a phase corrector, 15kHz lowpass filter, preemphasis network, and high-frequency limiter.

The lowpass filter prevents out-of-band components from affecting the operation of the high-frequency limiter and avoids intermodulation between out-of-band frequency components and in-band frequency components in the clipper.

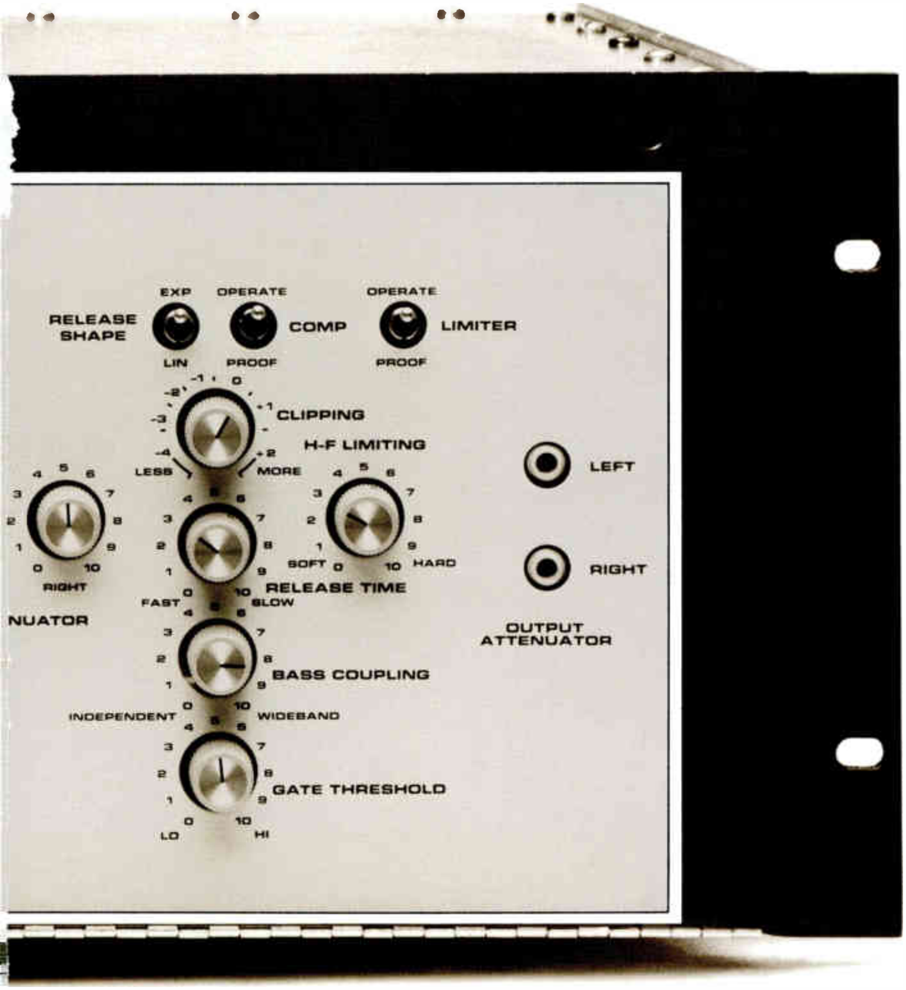
The high-frequency limiter is controlled by high frequencies *only* (rather than by the peak level of the preemphasized signal), eliminating any possibility of modulation of high frequency content by low frequency material. Its threshold of limiting is user-adjustable over a 3dB range, permitting brightness and high frequency distortion to be traded off according to your needs. Because the peak limiting system incorporates IM distortion cancellation, substantially more clipping can be accomplished without objectionable distortion than in conventional systems: Significantly improved high frequency power handling capability is achieved.

4. Loudness Controller: The concept of Loudness is different from the concept of Level. Loudness is *subjective* sound intensity: It is what the listener *perceives in his mind.* Level, on the other hand, can be measured in many objective ways: a VU meter and a PPM are two common level indicators in broadcast. No simple electrical measurement correlates well with loudness.

CBS Technology Center, in 20 years of experiments, has developed a technique of measuring loudness by means of a complex algorithm. This technique provides results which correlate well to loudness as subjectively judged by panels of listeners in extensive tests.

Ordinarily, gain reduction in OPTIMOD-TV is determined by the compressor's control circuitry. However, if the loudness exceeds a preset threshold, the Loudness Controller acts to further reduce the gain as necessary. This is the most advanced technique known for measuring and controlling the loudness of broadcast audio.

Because certain sounds in entertainment programming (pistol shots, explosions, or screeching tires, for example) are *supposed* to be loud for dramatic impact, the Loudness Controller can be turned on and off locally or by remote control.



SETUP CONTROLS
(BEHIND ACCESS DOOR)

(continued from inside)

5. "Hilbert-Transform Clipper":

The Hilbert-Transform Clipper provides the peak limiting function and contains filters to assure that the clipping does not introduce out-of-band frequency components above 19KHz. Unlike a conventional audio clipper, its action introduces no harmonic distortion when it processes frequency components below 4kHz. Simultaneously, IM distortion below 2.2kHz is sharply cancelled by an adaptation of the patented Orban feedforward distortion-cancelling filter.

The result is very low perceived distortion on both voice and music. Voice is most severely degraded by harmonic, not IM, distortion. No harmonic distortion is produced in the voice frequency range, keeping voice clean. Sibilance distortion is eliminated by the distortion-cancelling filter. In the frequency range in which music has substantial energy (particularly after preemphasis), IM distortion is minimized, optimizing music reproduction as well.

Because bandlimited voice (from 16mm optical film, for example) is so prevalent in TV audio and because bandlimited voice is exceedingly sensitive to the harmonic distortion introduced by more conventional clippers, the Hilbert-Transform clipper is extremely effective in achieving cleaner sound in day-to-day operations.

6. Frequency-Contoured Sidechain (FCS) Overshoot Corrector:

The output of the Hilbert-Transform Clipper contains overshoots due to the addition of the distortion-cancelling signal and to unavoidable overshoots in its integral 15kHz lowpass filter. These overshoots are eliminated in the FCS Overshoot Corrector without adding out-of-band frequency components: The circuit acts essentially as a "bandlimited safety clipper".

Because this circuit acts instanteously and employs no gain reduction or dynamic filtering, it causes neither pumping nor dulling of program material.

7. Noise Reduction Port, Output Amplifier, And Deemphasis:

At the output of the Overshoot Corrector, the signal is peak-controlled and pre-emphasized. The L and R outputs of the Overshoot Corrector are applied to a matrix which produces L + R and L – R. Jumpers determine whether the OPTIMOD-TV Noise Reduction Port is fed L/R or L + R/L – R signals.

The Noise Reduction encoder can be bypassed by the Noise Reduction IN/OUT switch on the rear panel of OPTIMOD-TV. The output of the switch (which selects either the input line to the encoder or the output line from the encoder) is applied to a balanced transformerless line amplifer with strappable deemphasis.

Best system peak control is obtained by defeating exciter preemphasis and applying the preemphasized signal from OPTIMOD-TV to the flat exciter. In some exciters it is inconvenient to defeat the preemphasis, so the exciter must be supplied with a "flat" (i.e., deemphasized) signal from OPTIMOD-TV, which is readily accomplished by moving jumpers.

The outputs of the line amplifiers are interfaced to the outside world through non-overshooting RFI filters effective from approximately 500kHz to 1GHz.

Summary:

OPTIMOD-TV is an integrated "system approach" to

- ☐ ride gain
- ☐ perform compression as desired
- ☐ control excessive loudness
- ☐ control peaks by high-frequency limiting, distortion-cancelling "Hilbert-transform clipping", and bandlimited overshoot correction.

This optimizes technical parameters to their practical limit while producing a sound at the viewer's ear which is perceived as natural, pleasant, and free from the processing artifacts that often plague other signal processing approaches.

Order Guide:

8182A	OPTIMOD-TV AUDIO PROCESSING SYSTEM
OPT-18	50us preemphasis installed
8182A/ST	Studio Accessory Chassis for "split" configuration
RET-25	Retrofit Kit to convert OPTIMOD-TV Model 8180A to Model 8182A. Requires return to the factory for modification and alignment.
MAN-8	Additional Copy Of 8182A Operating Manual

Enterprises inc
155 Executive Drive, Brookfield, WI 53005
(414) 782-7996 WI: (800) 772-7676

orban
Orban Associates Inc., 645 Bryant Street, San Francisco, CA 94107
Toll Free (800) 227-4498 In California (415) 957-1067 Telex 17-1480

Simplified System Description:

OPTIMOD-TV consists of seven basic blocks:

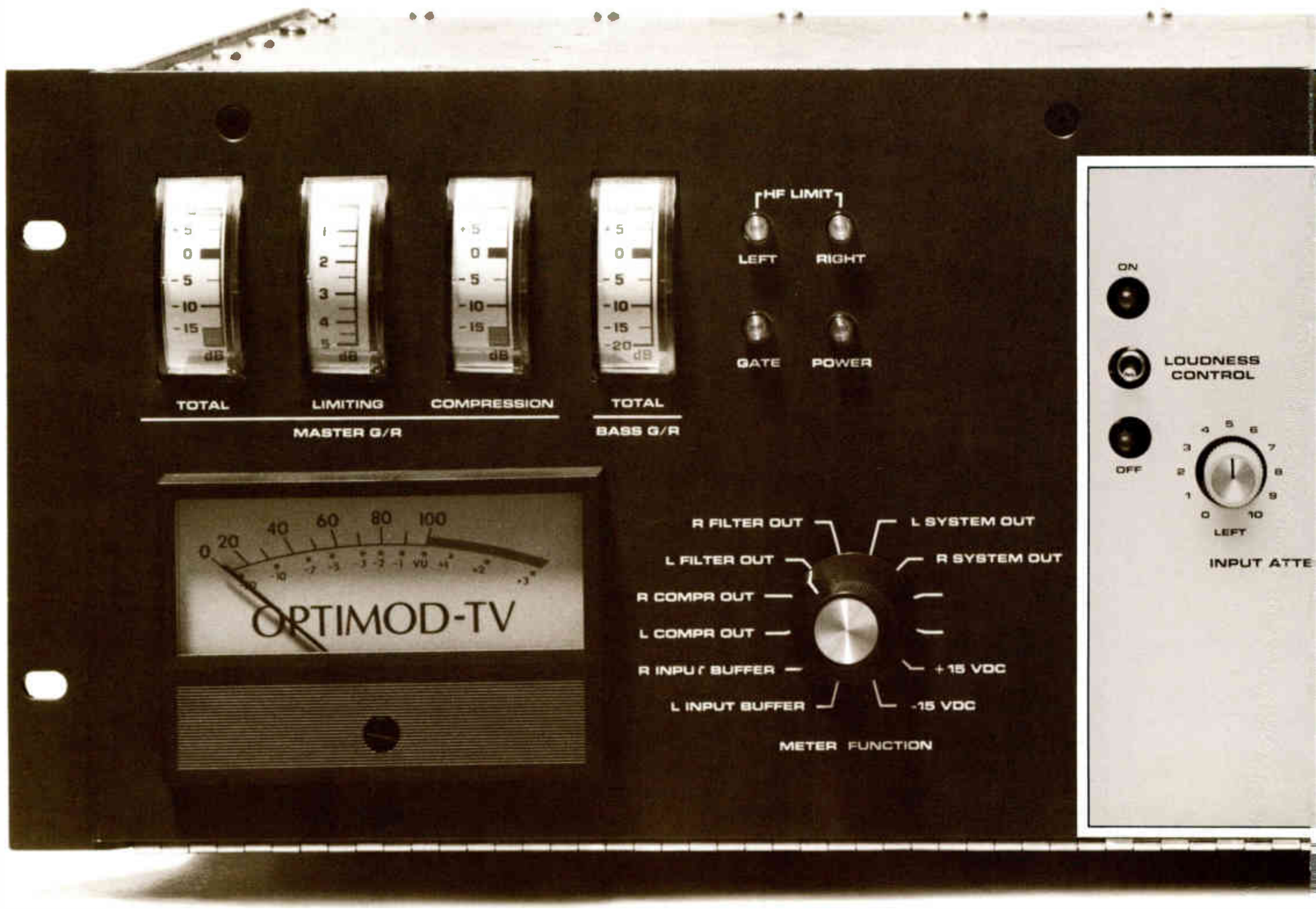
1. Input Conditioning Filter: An allpass phase scrambler to make peaks more symmetrical (thus reducing clipping distortion and permitting higher loudness), and a 30Hz 18dB/octave highpass filter to prevent subsonic information from disturbing the operation of the audio processing or exciters' AFC's.

2. Dual-Band Compressor: A "Bass" compressor which processes audio below 200Hz (12dB/octave crossover) in parallel with a "Master" compressor for the remainder. A BASS COUPLING control determines if the two bands operate discriminately ("independent" mode), or if the "Bass" band will be forced to track the "Master" band ("wideband" mode), preserving frequency balances. Intermediate bass coupling settings are also available.

Because of the unique design, the preemphasized output of the compressor can be directly applied to the peak limiting system: No further gain reduction is required for distortion control, and maximum naturalness is preserved.

Gating is provided to prevent noise rush-up during pauses (particularly with noisy 16mm optical sound tracks) and to make the 25dB gain reduction range more useful. The gating circuit is designed so that the gain does not get "stuck" forever in the 0 to 15dB gain reduction region—low-level program material is very slowly and impreceptibly increased in level even when gating is enabled.

The output level of the compressor is set by the CLIPPING control. This control thus sets the drive level to the subsequent high-frequency limiter and clipper, determining the amount of limiting and clipping.



Specifications

FREQUENCY RESPONSE

(System in PROOF mode)
Follows standard 75us preemphasis curve ±0.75dB, 50-15,000 Hz. (50us preemphasis available on special order.) Deemphasis jumper on line amplifier card permits flat output ±0.75dB 50-15,000Hz for use with external preemphasis. All preemphasis networks include a fourth-order lowpass filter and fourth-order phase corrector prior to the high-frequency limiter and clipper to prevent these elements from processing out-of-band program material and to minimize overshoot, thus minimizing the amount of high-frequency limiting and clipping.

INPUT CONDITIONING

Highpass Filter: Third-order Chebychev with 30Hz cutoff and 0.5dB passband ripple. Down 0.5dB at 30Hz; 10.5dB at 20Hz; 31.5dB at 10Hz. Protects against infrasonic destabilization of certain exciters' AFC's, as well as infrasonic gain modulation in the compressor.

Phase Scrambler: Allpass network makes peaks more symmetrical to best utilize the symmetrical peak overload characteristics of the FM medium.

NOISE

−75dB below 100% modulation, 50-15,000 Hz maximum; −81dB typical.

Total System Distortion (PROOF Mode; deemphasized; 100% Modulation)
Less than 0.25% THD, 50-15,000Hz (0.02% typical); less than 0.1% SMPTE Intermodulation Distortion (60/7000Hz; 4:1).
["THD" is defined as the root-sum-square (R.S.S.) sum of all harmonics, 50-30,000Hz. Noise (which is unavoidably included in the reading on a typical THD analyzer) is specifically excluded from this specification.]

"MASTER" BAND COMPRESSOR CHARACTERISTICS

Attack Time: approximately 1ms.

Release Time: program-controlled—varies according to program dynamics and amount of gain reduction (see text). Process can be scaled fast or slow by means of continuously variable RELEASE TIME control. Employs delayed release for distortion reduction.

Total Harmonic Distortion (measured at VCA output, OPERATE Mode, RELEASE TIME control centered): Less than 0.1%, 200-15,000Hz, +10 to −15dB gain reduction.

Available Gain Reduction: 25dB

Metering: Three dB-linear edgewise-reading gain reduction meters—
TOTAL is true peak-reading with electronic acceleration and peak-hold (+10 to −15dB);
LIMITING indicates fast peak limiting component of gain reduction (0-5dB);
COMPRESSION indicates slow compression component of gain reduction (+10 to −15dB).

Gain Control Element: True VCA. Proprietary Class-A design eliminates crossover notch distortion, modulation noise, and slewrate limiting found in competitive Class-AB designs.

"BASS" BAND COMPRESSOR CHARACTERISTICS

Attack Time: program-controlled; not adjustable.

Release Time: program-controlled; not adjustable. Incorporates delayed-release distortion reduction.

Total Harmonic Distortion (at VCA output, OPERATE mode):
Less than 0.1% THD, 50-200Hz, +10 to −20dB gain reduction.

Available Gain Reduction: 30dB

Metering: single dB-linear edgewise-reading gain reduction meter (+10 to −20dB).

Gain Reduction Element: Proprietary Class-A true VCA.

Bass Coupling (U.S. patent #4,249,042): Enables gain of "Bass' band to track gain of "Master" band to any degree, from identical tracking to fully independent operation. Adjustable with BASS COUPLING control.

CROSSOVER CHARACTERISTICS

Control: 6dB/octave @200Hz;

Program: 12dB/octave @200Hz in unique "distributed crossover" configuration (U.S. patent #4,249,042).

HIGH FREQUENCY LIMITER CHARACTERISTICS

Attack Time: approximately 5ms.

Release Time: approximately 20ms. Delayed release included for distortion reduction.

Mode: Left and right channels operate independently to avoid high frequencies in one channel causing audible timbre modulation of opposite channel.

Control Elements: Junction FET

Metering: Two LED's indicate HF limiting in L and R channels.

Threshold of HF Limiting: User-adjustable over 3dB range to meet format requirements.

HILBERT-TRANSFORM CLIPPER CHARACTERISTICS

Nominal Bandwidth: 15.4kHz

Distortion Characteristics: Less than 2.5% THD is produced by individual frequencies 30-4000Hz when driving the Hilbert-Transform Clipper to 6dB beyond its threshold of limiting. With drive frequencies above 4kHz, the characteristics revert to those of a very "hard" conventional clipper. Further distortion cancellation assures that, for any arbitrary input (including program material), distortion components in the frequency range from 0-2.2kHz are cancelled better than 30dB below overshoot compensator threshold (patent pending).

Delay Correction: Fourth-order allpass.

Amount of Clipping: User-adjustable over 6dB range to match format requirements.

FREQUENCY-CONTOURED SIDE CHAIN (FCS) OVERSHOOT COMPENSATOR CHARACTERISTICS (patent pending)

System Overshoot: The FCS circuit is best thought of as a "bandlimited safety clipper". It operates like a hard clipper, but does not produce out-of-band frequency components as a simple hard clipper would. Because the audio processing will sometimes limit steady-state material with high average energy (like sinewaves) or with very little high-frequency energy to levels below the threshold of clipping, it is difficult to state a clear and meaningful specification for the *system* overshoot performance of the FCS circuit.
The FCS circuit is followed by a safety clipper. The overshoot specification could be slightly improved if this safety clipper were set up to clip more frequently. However, the system is aligned at the factory such that the safety clipper is almost *never* active, thus fully preserving the bandlimiting provided by the FCS circuit. With this safety clipper alignment, the peak modulation will be controlled ±3.5% on arbitrary waveforms clipped to any degree by the FCS circuit (acting as a bandlimited safety clipper); peak modulation will not exceed this level on other material. With typical program material, peak modulation uncertainty is less than 2%.

Sinewave Modulation Ability: 93% modulation (i.e., 0.6dB below maximum overshoot level) at all sinewave frequencies, assuming sinewaves are applied to FCS input.

Dynamic Separation: better than 45dB.

Difference-Frequency Intermodulation: FCS circuit causes no more audible IM (such as sibilance splatter) than would a simple hard clipper clipping to the same depth. The entire OPTIMOD-TV processing system is specifically configured to prevent the FCS circuit from audibly degrading the difference-frequency distortion-cancellation properties of the earlier peak limiting system.

SYSTEM SEPARATION

Greater than 50dB, 50-15,000Hz; 60dB typical.

INPUT

Impedance: greater than 10K ohms, electronically balanced by means of true instrumentation amplifier. Requires balanced source ≤ 600ohms.

Common Mode Rejection: Greater than 60dB @60Hz.

Sensitivity: −10dBm produces 10dB "Master" Band gain reduction @1kHz. Removal of internal 20dB pad permits −30dBm to produce same effect.

Connector: Barrier strip (#6 screw).

OUTPUT

Source Impedance: 370 ohms, independent of OUTPUT ATTEN setting, balanced.

Recommended Load Impedance: 600 ohms ±20%.

Level: variable −infinity dBm to greater than +20dBm by means of 15-turn OUTPUT ATTEN controls.

Connector: Barrier strip (#6 screw). RF suppressed.

TEST JACKS (for Test use only)

Provides L and R lowpass filter output on RCA phono-type connectors on rear panel. Outputs are unbalanced.

OPERATING CONTROLS

VU Meter Selector: switches ASA-standard VU meter to read:

L or R Input Level	(L INPUT BUFFER)
L or R Compressor Output	(L COMPR OUT)
L or R Filter Out	(L FILTER)
L or R Line Amplifier Output	(L SYSTEM OUT)
±15 V Power Supply Voltages	

SETUP CONTROLS (front-panel, behind lockable swing-down door—see Fig. 4-5)

Compressor:
- Left and Right Input Attenuators
- "Master" Band Release Time
- Release Shape Switch
- Gate Threshold
- Bass Coupling
- Clipping
- High-Frequency Limiter Threshold

General:
- Left and Right Output Attenuators
- PROOF/OPERATE Switches (to defeat gain reduction, HF limiting, clipping and gating)
- Loudness Controller ON/OFF Switch
- Power ON/OFF Switch
- 115V/230V Selector Switch
- Noise Reduction IN/OUT Switch (rear panel)

POWER REQUIREMENT

115/230VAC, ±15%, 50-60Hz, approx. 31VA. IEC mains connector with detachable 3-wire "U-Ground" power cord supplied. Leakage to chassis less than 0.5mA. AC is RF-suppressed.

DIMENSIONS

19″ (48.3cm)W × 7″ (17.8cm) H × 12.5″ (31.2cm)D—4 rack units

ENVIRONMENTAL

Operating Temperature Range: 0-50°C (32-122°F).

Humidity: 0-95% R.H., non-condensing.

WARRANTY

One year, parts and labor. Subject to limitations set forth in our Standard Warranty.

All specifications subject to change without notice.

THE ORBAN 4000A TRANSMISSION LIMITER

Performance Highlights

- Ideal for transparently protecting transmission links (such as digital PCM, NICAM, analog microwave, and telephone/post lines) from overload.*
- Available in single-channel and dual-channel configurations. The dual-channel unit can be operated in stereo or as two independent units.
- Accurately and transparently limits levels without producing audible artifacts.
- Has very low static and dynamic distortion, thus producing extremely transparent, natural audio quality, both below and above threshold.
- Includes pre-emphasis limiting for five different pre-emphasis curves: 25µs, 50µs, 75µs, 150µs, and CCITT J.17.
- Rigorously limits its output bandwidth to 15kHz.
- Contains a built-in line-up tone generator for quick and accurate level setting in any system.
- Fully remote-controllable so that large facilities can perform routine network line-up checks centrally.
- 10-element LED bar-graph accurately indicates limiting.
- Equipped with a hard-wire relay bypass that can be activated locally or by remote control, and which activates automatically when the 4000A loses main power.
- Transformerless, balanced 10kΩ instrumentation-amplifier input. (An optional Jensen input transformer is available.)
- Transformerless, balanced, floating 30Ω output to ensure highest transparency and accurate pulse response. (An optional Jensen output transformer is available to protect against the very high common-mode voltages present on certain long lines.)
- Designed to meet all applicable international safety standards.

* However, the 4000A Transmission Limiter protects broadcast transmitters less effectively than Orban's OPTIMOD-AM, OPTIMOD-FM, OPTIMOD-TV, and OPTIMOD-HF—these OPTIMOD units should always be used wherever transmitter protection is required.

Ordering Details

4000A1 Transmission Limiter, one channel

4000A2 Transmission Limiter, two channel

Specify:

Mains voltage: 120V or 240V (mains voltage is user-switchable on the rear panel.)

Input Transformer: Optional

Output Transformer: Optional

TRANSMISSION LIMITING: WHY ORBAN'S APPROACH IS BETTER

Orban's 4000A Transmission Limiter is a uniquely transparent solution to the problem of protecting transmission systems from peak overload without side-effects or artifacts. A true "workhorse" product, its features precisely complement the requirements of the application, including easy set-up, and control features that greatly ease its use in large transmission networks.

It protects transmission links, such as digital links, post/telecom lines, microwave links, and satellite links, from overload. To achieve simple, error-free set-up and operation, the 4000A Transmission Limiter is configured with the minimum number of controls and indicators necessary to do its job properly. Unique, patented circuits and systems provide the most natural, uncolored sound available.

The Conventional Way

Traditionally, transmission limiters have used peak-sensing automatic gain control (AGC) circuits to control peak levels. Superficially, this approach seems reasonable—the purpose of a transmission limiter is to control the peak levels in a transmission channel.

This approach ignores one crucial requirement for transmission limiter performance: the limiter must provide ***natural-sounding*** control that is ***undetectable to the ear*** except by an A/B comparison to the original source material.

Because the human ear is basically average-sensing, not peak-sensing, the simplistic peak-sensing AGC technique causes highly unnatural variations in perceived loudness. Material with a high peak-to-average ratio (such as voice) emerges from such a limiter much quieter than material with a low peak-to-average ratio (such as some recorded music). The ear perceives this as an unnatural, unpleasant "pumping" quality. Thus the peak-sensing AGC limiter fails to provide natural sound quality, and we must use more sophisticated techniques.

To achieve natural sound quality, the gain control section of the limiter must respond like the ear. This means that the gain control must respond approximately to the power (not the peak level) in the signal. Further, because the sensitivity of the ear decreases dramatically below 150Hz, the control must be frequency-weighted to compensate. Otherwise, heavy bass would audibly modulate the loudness of midrange program material—a problem called "spectral gain intermodulation."

Orban's Way

To prevent spectral gain intermodulation, the gain control in the Orban 4000A occurs in two frequency bands—above and below 150Hz. Further, the control loop level detector has an attack time of approximately 2 milliseconds, so it approximates a power response instead of a peak response.

Because its two frequency bands are cross-coupled, the 4000A ordinarily behaves as a wide-band limiter and preserves the frequency balance of the input program material. However, if material having very heavy bass appears, the "bass" band (below 150Hz) will temporarily produce more limiting than the "master" band (above 150Hz). This prevents the bass from audibly modulating the loudness of the midrange.

Sophisticated Peak Control

Because the gain control section of the 4000A is not peak-sensing, its output contains peak overshoots that must be eliminated by further circuitry. The 4000A provides three cascaded curcuits to control peaks: the high-frequency limiter, the "Smart Clipper™" distortion-canceling clipper, and the Frequency-Contoured Sidechain™ overshoot corrector.

The high-frequency limiter is a program-controlled dynamic filter that temporarily rolls off excessive high frequency power (caused by pre-emphasis) to prevent distortion in the following clipper circuitry. It is used when the 4000A is configured to control peak levels in a pre-emphasized transmission link.

Traditionally, clippers cause objectionable distortion. Such distortion is prevented in the Transmission Limiter's "Smart Clipper" by proprietary, patented circuitry that analyzes the frequency spectrum of the distortion products produced by clipping, and manipulates this distortion spectrum to ensure that the distortion products are psychoacoustically masked by the desired program material.

Such manipulation of the distortion spectrum introduces a small amount of peak overshoot into the output of the "Smart Clipper." Further, this circuit contains low-pass filters that strictly constrain the output spectrum to 15kHz, but which overshoot. Orban's patented Frequency-Contoured Sidechain overshoot corrector eliminates these overshoots. This circuit derives a band-limited signal that can be added to its input signal to cancel overshoots without destroying the spectral integrity of the signal, as simple clipping would.

TRANSMISSION LIMITING CONTINUED

Delay-Line Techniques vs. Distortion-canceling Clipping

The 4000A Transmission Limiter achieves maximally transparent sound below threshold and extremely natural dynamics above threshold. Our goal was to have the transition into limiting undetectable to the ear. We feel that delay line techniques are incongruent with these goals because a delay-line limiter is simply a highly refined "peak-sensing AGC circuit."

While a delay-line limiter can achieve very low perceived distortion, it does so at the expense of having an extremely fast attack time such that limiting is produced on every transient overshoot, no matter how brief. This effect is somewhat reduced by an "automatic" release circuit, but the inevitable consequence is that the average power at the output of the limiter is strongly influenced by the peak-to-average ratio of the program material. Thus material with an unusually high peak-to-average ratio can unnaturally reduce the average power. Conversely, material with an unusually low peak-to-average ratio can be amplified to unnaturally loud levels. The overall subjective effect is that changes in the program waveform produce somewhat unnatural dynamic variations; the sound of the limiter is not "effortless."

High-Frequency Limiting for Minimum Perceived HF Loss

A limiter that does not use clipping to control peaks must control pre-emphasis-induced overshoots with a fast peak-sensing variable-emphasis limiter. Such limiters tend to cause severe audible dulling of certain program material.

The 4000A uses a HF limiter which is designed ***only to prevent audible distortion in the following distortion-canceling clipper***. The distortion-canceling clipper does almost all of the work in limiting HF peaks. Because it operates on ***each individual peak*** without affecting the peak's neighbors, the distortion-canceling clipper causes far less audible HF loss than does a traditional variable-emphasis limiter.

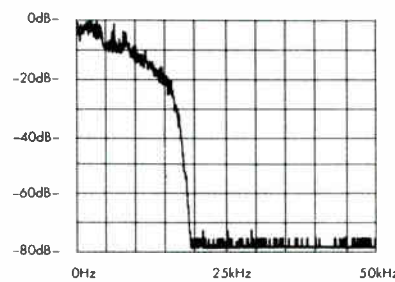


Figure 1: Power spectral density at the 4000A's Output using "maximum peak hold" measurement. (10kHz/div horizontal; 10dB/div vertical)

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TV Broadcast Products
USA Suggested List Prices

Effective: 1 August 1990

OPTIMOD-TV

		USA LIST PRICE
AUDIO PROCESSOR		
8182A/U75	OPTIMOD-TV Audio Processor 120V 75µs (37lb/17kg)	$5,900.00
STUDIO CHASSIS		
8182AST/U	OPTIMOD-TV Studio Chassis 120V (16lb/8kg)	1,000.00
STEREO GENERATOR		
8185A/U	BTSC TV Stereo Generator 120V (34lb/16kg)	7,200.00
SECOND AUDIO PROGRAM (SAP) GENERATOR		
8182ASAP/U	BTSC SAP Generator with Monitor Card 120V (39lb/18kg)	5,500.00
PRO CHANNEL GENERATORS		
8182APRO	BTSC PRO Generator for 8182A/SG (2lb/1kg)	1,300.00
8185APRO	BTSC PRO Generator for 8185A (2lb/1kg)	1,300.00
TV ACCESSORIES		
ACC021	dbx Monitor Card for earlier 8182A/SAP (2lb/1kg)	700.00
RET037	8180A Connector Upgrade to accept SG (2lb/1kg)	65.00
RET025	8180A to 8182A Factory Upgrade (37lb/17kg)	1,000.00
RET026	8180A/ST to 8182A/ST Factory Upgrade (16lb/8kg)	300.00
STEREO SYNTHESIZER		
275A	Automatic Stereo Synthesizer (11lb/5kg)	2,295.00
275ARC	Remote Control Panel for 275A (4lb/2kg)	295.00

TRANSMISSION LIMITER

TRANSMISSION LIMITERS

4000A1/U75	Transmission Limiter, 1ch 120V 75µs (14lb/7kg)	1,900.00
4000A1/UT75	Transmission Limiter, 1ch 120V OPTx 75µs (14lb/7kg)	2,050.00
4000A1/UTT75	Transmission Limiter, 1ch 120V IPTx OPTx 75µs (14lb/7kg)	2,130.00
4000A2/U75	Transmission Limiter, 2ch 120V 75µs (17lb/8kg)	2,500.00
4000A2/UT75	Transmission Limiter, 2ch 120V OPTx 75µs (17lb/8kg)	2,800.00
4000A2/UTT75	Transmission Limiter, 2ch 120V IPTx OPTx 75µs (17lb/8kg)	2,960.00

803008-001-11 6/90

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PRICES:	FOB AKG Acoustics, Inc., San Leandro, CA, USA. Prices are subject to change without notice.
TERMS:	6% 30 days/net 31 days, to approved open account customers. 6% cash before delivery (CBD). 6% prompt payment discount is on the net price of goods, excluding freight or insurance.
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WIRE TRANSFER:	Wire funds through the United States Federal Reserve Bank, in favor of Wells Fargo Bank ABA 121000248, to credit the account of AKG Acoustics, Inc. No. 4121086508.

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DIFFERENCES BETWEEN THE 4000A AND OPTIMOD-FM

The main AGC stage of the 4000A is a gated, dual-band limiter somewhat similar to the one used in Orban's OPTIMOD-FM. However, several changes were made to make the circuit more suited to the transmission limiter function. The consequences of these changes are:

- The overall frequency response of the 4000A is typically ±0.25dB, 30–15,000Hz: slightly flatter than that of OPTIMOD-FM.
- The crossover has constant group delay after the bands are summed; there is no phase rotation. We felt that this was necessary to achieve highest audible transparency.
- The overall group delay through the entire 4000A system (configured with FLAT output) is virtually constant: 359µs ±5%, 30–15,000Hz.
- The two bands in the 4000A's AGC are cross-coupled according to Orban's patented scheme so that the frequency response is almost always flat (excluding the effects of any HF limiting). Only with material having *extreme* bass does the bass band produce extra limiting to prevent audible spectral gain intermodulation between the bass and midrange.
- The 4000A's control circuit is tuned to produce little or no increase of program density. There is no need for a transmission limiter to significantly increase program density, and the 4000A does not do so. Instead, it produces the most natural, subtle action that we can achieve.
- The static "knee" of the 4000A's compression curve is slightly "harder" than that of OPTIMOD-FM. This makes it easier to operate the 4000A below threshold most of the time, because the transition into limiting is better-defined.
- The 4000A controls its output spectrum significantly better than a standard OPTIMOD-FM; spectral control has been traded-off against overshoot control, which is slightly poorer than OPTIMOD-FM (by about 1dB). Fig.1 shows the output power spectrum as measured by the stringent "maximum peak hold" technique with an 801-line FFT dynamic signal analyzer (Hewlett-Packard 3562A).
- The 4000A can be configured to provide full peak control for CCITT J.17 pre-emphasis. Further, its HF limiter can be easily jumpered to protect J.17, 150µs, 75µs, 50µs, or 25µs systems.
- The 4000A is a transparent *transmission limiter* designed for *transmission link protection*. Its range of limiting has been limited to 15dB. It cannot accept the 8100A/XT2 Six-Band Limiter Accessory Chassis, nor can it be operated so that its low-frequency and high-frequency bands are independent. There is no RELEASE TIME control. The consequence of all this is that the 4000A cannot create a "competitively processed" sound with pop music. OPTIMOD-FM is the ideal choice for this function.

APPLICATION

Location of 4000A in System

The 4000A is usually installed immediately prior to a transmission link such as a land, PTT, or Telecom line, digital link, or microwave link. To achieve best peak control, it is essential not to insert anything between the output of the 4000A and the input of the transmission link that would change the *shape* of the tightly peak-controlled output of the 4000A.

Specification for the Output Link

Anything that affects the constancy of the frequency response or the group delay between the 4000A's output and the input of the driven transmission link must be qualified to ensure that its frequency response is ±0.25dB from 30–15,000Hz, and that its deviation from linear phase does not exceed ±10° over this frequency range. The phase specification implies that the low-frequency response limit must be 0.15Hz or lower (at –3dB) unless special group delay equalization is used at low frequencies. Low-pass filters (including anti-aliasing filters in digital links), high-pass filters, transformers, distribution amplifiers, and long transmission lines are all suspect, and must be tested and qualified.

Digital Links

The stopband of the 4000A's filtering begins at approximately 18.5kHz. At this frequency the power spectrum is greater than 75dB below 100% modulation, as measured by the extremely stringent "maximum peak hold" technique using an 801-line FFT analyzer. This is slightly beyond the Nyquist frequency of 16kHz in an EBU-standard 32kHz link. However, the spectrum falls very rapidly beyond 16kHz (see Fig.1). In FM stereo applications, material above 15kHz will be attenuated by the 15kHz low-pass filter in the FM stereo encoder. In a 32kHz sample-rate transmission link, 17kHz will alias to 15kHz. Thus, only energy beyond 17kHz could cause trouble in an FM stereo system. Because the power spectrum at the 4000A's output is already down at least 40dB at 17kHz, we believe that the system is adequately protected from audible aliasing by the 4000A's filtering alone.

This has a significant advantage. Because the output of the 4000A can be strapped to be either "flat" or "pre-emphasized," the 4000A can provide *both* the J.17 pre-emphasis and anti-aliasing filtering functions prior to a PCM, NICAM or similar digital transmission link. The pre-emphasis filters and linear anti-aliasing filters with which the links are presently fitted can thus be removed. Since these elements overshoot, their removal eliminates the overshoot, permitting the average modulation of the link (and thus, its signal-to-noise ratio) to be improved by several decibels without compromising the subjective quality of the audio.

Microwave Links

It is usually easy to modify a microwave link to meet the specification for frequency response and phase linearity stated above. Many such links have been designed to be easily configured at the factory for "composite" operation, where the entire FM stereo baseband is passed, including the pilot tone and stereo subchannel. The requirements for maintaining stereo separation in "composite" operation are similar to the requirements for high waveform fidelity with low overshoot. Therefore, most links have the *potential* for excellent waveform fidelity if they are configured for "composite" operation (even if a "composite" FM stereo signal is not actually being applied to the link).

The 4000A can provide the necessary pre-emphasis, low-pass filtering, and peak control to optimally drive such a microwave transmitter. All audio low-pass filters and pre-emphasis filters in the microwave transmitter should be bypassed or removed (as they would be when the transmitter was configured for "composite" operation).

Telephone or Post Lines

Most such lines have poor low-frequency and high-frequency phase linearity. When the 4000A drives such lines, a properly peak-controlled signal is unlikely to emerge at the receive end. We recommend following traditional line-up procedures with such lines; although the 4000A's uniquely effective peak control is less significant in this application, its natural, transparent sound quality still makes it the protection limiter of choice.

Subcarriers

Sometimes subcarriers are used as transmission links, particularly in television, where several aural subcarriers may be available on a video microwave STL.

FM-Modulated Subcarriers: The 4000A can provide the necessary pre-emphasis, low-pass filtering, and peak control to optimally drive an FM subcarrier generator. All audio low-pass filters and pre-emphasis filters in the subcarrier generator should be bypassed or removed. The 4000A should be strapped to generate the same pre-emphasis curve as the bypassed pre-emphasis filter.

Single-Sideband Companded Subcarriers: The peak modulation levels entering a single-sideband companded subcarrier generator are greatly changed by the noise reduction compressor and by the single-sideband modulator. Therefore, you must leave headroom to accommodate the unavoidable overshoots generated by these elements. You must experimentally determine the input drive level necessary to produce correct modulation. Do this by adjusting the 4000A's OUTPUT ATTENUATOR until the subcarrier modulation (as read on the appropriate monitoring instrument) is correct.

Easy System Line-Up

To facilitate system line-up, the 4000A has a special TEST mode that raises the threshold of the gain control section to 100% peak modulation, permitting system line-up tones to be passed without limiting. The 4000A also has a built-in 400Hz oscillator that will produce a tone at 100% peak modulation.

Specifications

Performance

Frequency response (20–15,000Hz): ±0.25dB below leveler, compressor, and high-frequency limiter thresholds.

RMS noise: >87dB (90dB typical) below 100% modulation with high-frequency limiter strapped for flat output. Measured in a 20–20,000Hz bandwidth with a true-R.M.S. meter.

Total harmonic distortion (TEST mode): <0.05%, 20–15,000Hz. Measured at a level equivalent to 90% peak modulation.

Total harmonic distortion (OPERATE mode): <0.05% at 1kHz. Typically <0.2% at 20Hz, <0.1% at 100Hz, <0.05% at 200–15,000Hz. Measured with 5dB of limiting at 1kHz. (As with any limiter, low-frequency distortion will rise as limiting is increased because the limiter acts on each cycle of the low-frequency waveform.)

SMPTE Intermodulation Distortion (TEST mode): <0.075%. Measured at level equivalent to 90% peak modulation; 60/7000Hz; 4:1.

SMPTE Intermodulation Distortion (OPERATE mode): <0.25%. Measured with 5dB of limiting.

Interchannel crosstalk (dual-channel 4000A/2 unit): Better than –70dB at 15kHz (–75dB typical), falling at 6dB/octave below 15kHz.

Overshoot: 1dB maximum (referred to the low-frequency "clipping threshold" of the FCS Overshoot Compensator).

Spectral Control: see Fig. 1 on page 2.

Installation

Audio Input

Impedance: >10kΩ, active balanced, EMI-suppressed. Input transformer option available.

Operating level: Usable with –30dBu to +10dBu lines. (0dBu = 0.775V RMS)

Connectors: XLR connectors.

Audio Output

Impedance: 30Ω, electronically balanced and floating to simulate true transformer output. Minimum load impedance is 600Ω. Output can be unbalanced by grounding one output terminal. Output transformer option available.

Level: Front-panel controls permit use with –10dBm to +8dBm systems. Output clipping level is > +20dBm @600Ω.

Connectors: XLR connectors.

Physical

Controls: Momentary, pushbutton.

Meter: 10-segment LED bargraph display shows limiting, 0–15dB.

Indicators: Two LEDs light to show operation of gating and high-frequency limiting.

Dimensions: 19" (48.3 cm) wide, 11 1/4" (28.6 cm) deep, 3 1/2" (8.9 cm) high.

Operating temperature range: 32–113°F (0–45°C).

Power requirements: 115/230 volts AC ±10%, 50–60Hz, 16VA. IEC mains connector with detachable three-wire "U-Ground" power cord and plug supplied. EMI-suppressed. AC power input is RFI-suppressed.

Fuse: 1/2-A 3AG 250V slow-blow for 115V operation; 1/4-A 3AG 250V slow-blow for 230V operation (or 5 x 20mm Type T with adapter).

Options

Security cover (acrylic): To prevent unauthorized adjustment of controls. Order ACC–012CL for a clear cover, ACC–012BL for a transparent blue cover, or ACC–012WH for an opaque white cover.

Audio Processing Circuitry

Dual-Band Limiter

Attack time: Approximately 2ms; program-dependent.

Release time: Program-dependent; not user-adjustable.

Compression ratio: >20:1 (static); program-dependent (dynamic).

Range of limiting: 15dB.

Interchannel tracking: ±0.5dB (dual-channel 4000A/2 strapped for COUPLED operation).

Total harmonic distortion (TEST mode): <0.035%, 20–15,000Hz. Measured at level equivalent to 90% peak modulation.

SMPTE intermodulation distortion: <0.075% Measured at level equivalent to 90% peak modulation.

Limiting element: Class-A VCA.

High-frequency Limiter

Pre-emphasis: Five pre-emphasis curves: 25µs, 50µs, 75µs, 150µs, and CCITT J.17. Can be strapped for flat or pre-emphasized output.

Response: The high-frequency limiting threshold and attack time have been set so that no audible distortion is produced with dynamic program material that has been processed by the leveler/compressor, smart clipper, and FES overshoot compensator. Because these settings have taken into account the peak-to-average ratio of the leveler/compressor's output, it is not possible to specify the high-frequency limiter's response to test tones with simple, meaningful numbers.

Total harmonic distortion: The high-frequency limiter/clipper will add no more than 0.02% THD to sine wave test tones that have been processed by the leveler/compressor.

Release time: Approximately 30ms, program-dependent.

Limiting element: Junction FET.

HF limiting curve: Shelving, 6dB/octave.

"Smart Clipper"

Distortion: Below its clipping threshold, the "Smart Clipper" will add no more than 0.025% THD to sine wave test tones that have been processed by the previous circuitry. Above threshold the circuitry cancels clipping-induced intermodulation distortion in a complex, frequency-dependent manner to maximize psycho-acoustic masking of such distortion, minimizing its perceptibility.

FCS Overshoot Compensator

Distortion: Below its clipping threshold, the FCS Overshoot Compensator will add no more than 0.01% THD to sine wave test tones that have been processed by the previous circuitry. Above threshold the circuitry performs "band-limited clipping" to remove overshoot without introducing out-of-band distortion power above 15kHz.

Remote Control: TONE, TEST, OPERATE and BYPASS moves can be selected by remote control. The optically-isolated remote control inputs will operate from 6–48V, AC or DC, momentary.

Warranty

One year, parts and labor. Subject to the limitations set forth in Orban's Standard Warranty Agreement.

Specifications subject to change without notice.

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800068–000–02 2K–CC–8/90

control for the subcarrier, the CO-OPERATOR can also provide the rarely found 150us HF limiting function, which permits the most efficient use of the SCA channel by eliminating overmodulation on HF peaks.

Circuit Features

The CO-OPERATOR consists of a subtle AGC/compressor cascaded with a HF limiter. To provide absolute peak control without the pumping associated with fast-attack limiter circuits, the HF limiter circuit includes a pre-emphasized clipper (which is jumper-defeatable). When the HF LIMITER switch is set to OUT, the entire HF limiter circuit is removed from the signal path, ensuring maximum transparency.

Although AGC attack and release times adapt automatically to the nature of the program material to ensure the smoothest and least audible compression, the operator can fine-tune the CO-OPERATOR to obtain the precise type of dynamic range control desired by adjusting the relative speed of the program-controlled release process. A faster "compression" function can be switched in to provide additional transient overshoot protection for material with abrupt level changes or unusually high peak-to-average ratios.

The defeatable silence gate (controlled by the front-panel GATE THRESHOLD control) avoids noise rush-up during low-level program material by radically slowing the release process, thereby forcing the VCA gain to move slowly toward the average of the last 30 seconds of gain reduction. (This is an "automatic idle gain" function, similar to the manually-set IDLE GAIN on our 424A Gated Compressor/Limiter/De-Esser.)

VCA noise in the gain reduction circuitry (typically –90dB below the VCA output clipping point, 20-20kHz unweighted) stays virtually constant as the VCA gain changes. This results in very low noise under real-life conditions. The VCA used is a proprietary Orban Class-A VCA with very low static distortion.

Summary

The Orban CO-OPERATOR provides an unprecedented combination of versatility, audio quality, and ease-of-use. Like any such device, specifications and brochures can't really describe what it sounds like or how it feels to use it. We invite you to try the CO-OPERATOR for yourself. We think you will find it to be a talented and indispensable assistant.

Projected delivery, first unit: January, 1987 Estimated price: $959

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PRELIMINARY BROCHURE

The Orban CO-OPERATOR™

An Integrated, Easy-to-Use Gated Stereo Leveler/Compressor/HF Limiter

Transparent level control, compression, and high-frequency limiting in a powerful, compact, and economical dual-channel package. Automatically rides gain, controls excessive high-frequency levels, and limits peaks in the radio, teleproduction, or recording studio. Protects recording tape, broadcast cart machines, microwave links, FM SCAs, cassette masters, and sound systems.

- A defeatable "silence gate" prevents noise rush-up, holes, pumping, and breathing by inhibiting sudden gain increases once the signal level falls below a preset threshold (during pauses or low-level program material).
- Six switchable HF limiter curves (25 to 150us) match the HF limiting to the medium or device being protected and optimize control of excessive sibilance.
- A defeatable clipper follows the HF limiter, so the unit can be used for absolute peak protection.
- The gain compression recovery rate is switch-selectable. HARD KNEE recovery (at a constant rate) is best for single tracks and live voice. The SOFT KNEE recovery rate (which slows during recovery) provides more subtle gain-riding for mixed program material.
- A faster "compression" function can be switched-in to provide additional transient overshoot protection for material with abrupt level changes or unusually high peak-to-average ratios.
- Switchable for stereo-tracking or independent two-channel operation.
- The least-used controls are concealed behind a security panel to avoid confusing non-technical operators, and to permit tamper-proof calibration.
- Two LED bargraphs per channel simultaneously display gain reduction and peak output level.
- Output level meter can be calibrated to match the overload point of the device being driven.
- Balanced, floating inputs and outputs are EMI-suppressed.
- 25dB gain reduction range is achieved with a low-distortion, Class-A VCA.
- Two channels in a space-saving, rugged, all-metal, 1-3/4" package.
- Hard-wired bypass switch included.

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Main: (415) 957-1063 Sales: (415) 957-1067 Telex: 17-1480

Your Assistant Operator

The Orban CO-OPERATOR is a friendly, automatic "assistant operator" that smoothly and unobtrusively rides gain and limits peaks.

The defeatable HF limiter can be used to prevent pre-emphasized material from overloading tape recorders, disk cutters, sound systems, broadcast STLs, FM SCAs, and cassette masters with excessive high-frequency energy. Controlling high-frequency energy permits average recording or transmission levels to be increased, yielding improved signal-to-noise ratios.

As a result of Orban's years of research and practical experience in the broadcast signal processing field, the CO-OPERATOR has outstandingly transparent audio performance. This performance is achieved through the use of finely-tuned control loops to eliminate the dynamic distortions that are the downfall of most conventional compressors and limiters, and by the use of a clean, Class-A VCA to ensure negligible static distortion and noise.

After audio quality, ease of use was the highest priority in the development of the CO-OPERATOR. The operator need only be concerned with three controls:

INPUT ATTENUATOR	Determines the amount of gain reduction.
GATE THRESHOLD	Determines the level below which ordinary automatic gain control (AGC) action is "frozen" (to prevent noise rush-up during pauses and low-level program).
RELEASE TIME	Speeds or slows the program-controlled release time to match the processing to the flavor of the audio being processed.

The operator can also select the gain reduction recovery rate (HARD KNEE or SOFT KNEE leveling) and additional transient overload protection, according to the demands of the material. Easily interpreted, *independent* LED bargraphs display gain reduction and output levels.

Versatility Without Confusion

The CO-OPERATOR offers unprecedented flexibility to effectively control levels in a wide range of professional applications. To ensure ease of use and to avoid confusing the less technical operator, set-up controls and jumpers are concealed.

The controls which calibrate the OUTPUT LEVEL meters and those which determine the actual output level are behind a pop-off door on the front panel, as are the switches which determine the pre-emphasis curves of the high-frequency limiters. The clippers which follow the HF limiter are strapped in or out with a jumper on the circuit board. Thus, once the CO-OPERATOR has been set up by the installing engineer, these controls are safely out of sight. (Where the CO-OPERATOR protects an STL, SCA, or transmitter, an optional Security Cover can prevent unauthorized adjustment of *any* control.)

Applications

Recording Studios: The outstandingly transparent CO-OPERATOR is ideally suited for subtly reducing the dynamic range of an entire mix with the KNEE control set to SOFT. The HARD KNEE setting provides effective gain riding on single tracks, increasing "punch" and intelligibility while retaining the basic feel of the performance and the apparent dynamic range of the voice or instrument. The RELEASE TIME control (a feature absent from other "invisible" compressors) is invaluable for governing the uniformity of loudness in the final result. Compact disks mercilessly reveal the side-effects of conventional compressors; the CO-OPERATOR's freedom from dynamic distortion meets the challenge of the digital age.

Audio and Video Production: In these fast-moving worlds, time is money. The CO-OPERATOR can be set up quickly and easily to protect VTRs, ATRs, or cart machines from overload. Use the CO-OPERATOR on mike channels -- its smoothness and silence-gating guarantee uniform, punchy voice quality without noise rush-up during pauses.

Cassette Duplication: Even with the latest technologies like "hot" tape and Dolby® HX Pro, there are still some masters which can cause high-frequency saturation in cassette dupes. The CO-OPERATOR's high-frequency limiter can be set to eliminate problems caused by synthesizers, cymbals, and sibilance, while still permitting high average modulation on the cassette. The broadband AGC can be defeated to permit use of the HF limiter alone when automatic gain riding or broadband peak protection is not desired.

Sound Systems: The CO-OPERATOR can protect your high-frequency drivers and power amplifiers from overload more effectively than any other available level control devices. And its natural control action will not otherwise "color" your system. LED bargraph displays make visual reference in low-light situations a snap.

Broadcast Studio/Transmitter Link Protection: The STL is often the weak link in the broadcast chain due to barely adequate signal-to-noise ratio. Clean protective limiting ahead of the pre-emphasized STL transmitter maximizes the signal-to-noise ratio at the receiver and prevents overload.

FM Subcarriers: Voice or music SCA channels are typically pre-emphasized at 150us. In addition to providing transparent gain riding and level

Specifications

Input

Impedance: greater than 10 k ohms active balanced (interfaces with balanced or unbalanced sources).
Level: –15dBm produces 10dB gain reduction with ATTACK TIME control centered, INPUT ATTEN control fully CW, and RATIO control at infinity-to-one.
Absolute Overload Level: +21dBm

Output

Impedance: approximately 100 ohms, electronically balanced and floating (drives balanced or unbalanced loads)
Levels: +4dBm nominal; absolute peak overload better than +19dBm

Frequency Response

±0.25dB 20-20,000 Hz below limiting threshold

Compressor/Limiter Characteristics

Attack Time: manually adjustable in range of approximately 500us to 200ms; automatically scaled by program content.
Release Time: adjustable in range of approximately 3dB/sec to 80dB/sec; automatically scaled by program content.

Compression Ratio: adjustable from 2:1 to infinity-to-one at threshold. Lower ratios automatically increase as gain reduction increases.
Range of Gain Reduction: from 15dB to 35dB depending on setting of THRESHOLD control; 25dB with THRESHOLD control at center detent.
Total Harmonic Distortion (ATTACK and RELEASE TIME controls centered, infinite RATIO, 15dB gain reduction): less than 0.05% @ 1kHz. Typically 0.04% @ 100Hz; 0.025% @ 1kHz; 0.035% @ 10kHz.
SMPTE IM Distortion (controls set as above; 60/7000Hz 4:1; 15dB gain reduction): typically 0.2%.
Tracking of Multiple Channels: ±0.5dB.

System Noise

RMS noise in 20-20kHz bandwidth better than 85dB below output clipping threshold for any degree of gain reduction; 90dB typical.

Crosstalk (414A Only)

Better than –70dB @ 20kHz. Unmeasureable below 10kHz.

Operating Controls

INPUT ATTENUATOR
OUTPUT ATTENUATOR
THRESHOLD
COMPRESSION RATIO
ATTACK TIME
RELEASE TIME
SYSTEM OPERATE/BYPASS (Hardwire Bypass)
STEREO COUPLING (414A Only)
POWER ON/OFF

Indicators

GAIN REDUCTION METER
GAIN REDUCTION OVERLOAD LAMP
OUTPUT CLIP LAMP

Power Requirement

115/230 VAC ±10%; 50-60Hz.
U-ground power cord attached.

Dimensions

412A
19" (48.3cm) wide × 1.75" (4.5cm/1 unit) high × 5.3" (13.5cm) deep
414A
19" (48.3cm) wide × 3.5" (8.9cm/2 units) high × 5.3" (13.5cm) deep

Operating Temperature

0-45° C

Warranty

One year, parts and labor. Subject to limitations set forth in our Standard Warranty Agreement.

SPECIFICATIONS SUBJECT TO CHANGE WITHOUT NOTICE.

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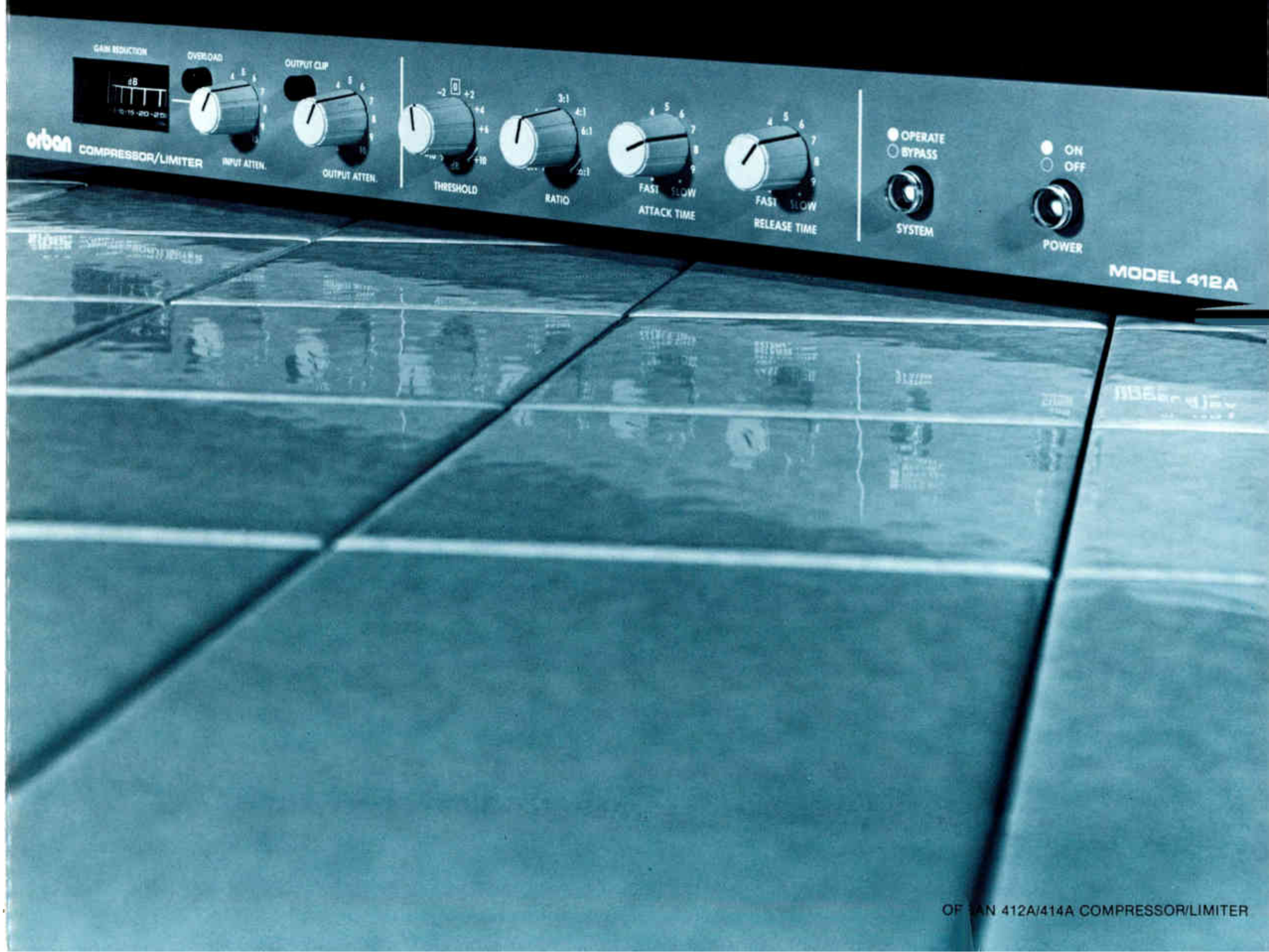
The Orban 412A Compressor/Limiter

The Essential AGC: A basic, cost-effective compressor/limiter with remarkably natural sound and extraordinary ease-of-use.



Performance Highlights

- ☐ Streamlined, straightforward front panel offers the most-demanded user controls, including ATTACK TIME, RELEASE TIME, RATIO, and THRESHOLD. These wide-range controls permit extremely natural sound or special effects.
- ☐ Exclusive Orban feedback control circuitry (adapted from our popular 424A Gated Compressor/Limiter/De-Esser) achieves remarkably transparent sound.
- ☐ User controls interact intelligently to simplify and speed setup, and to prevent errors.
- ☐ Peak limiting and compressor functions are crosscoupled to eliminate potential pumping and modulation effects.
- ☐ THRESHOLD control with 20dB range allows user to determine the level at which gain reduction first occurs, without changing below-threshold gain. Ideal for sound reinforcement applications.
- ☐ Proprietary circuitry achieves optimum headroom and signal-to-noise regardless of THRESHOLD control setting.
- ☐ Front-panel OUTPUT ATTENUATOR control with OUTPUT CLIP LED to indicate line amplifier clipping.
- ☐ Illuminated, true peak-reading GAIN REDUCTION meter is more accurate and readable than LED displays.
- ☐ GAIN REDUCTION OVERLOAD lamp warns of control circuit overload due to a demand for G/R which exceeds the range of the VCA.
- ☐ Hard-wired system bypass switch for fail-safe protection.
- ☐ Side-chain externally accessible for special effects such as frequency-selective limiting.
- ☐ Proprietary Class-A Orban VCA features very low distortion and noise.
- ☐ Stereo 414A has STEREO COUPLING switch to permit either stereo or dual-mono operation; an unlimited number of units can be wire-coupled to track ±0.5dB.
- ☐ All-metal chassis with RFI suppression on input, output, and AC leads.

Back to Basics

The 412A is Orban's entry into the general-purpose compressor/limiter sweepstakes—it's designed to make **you**, the audio professional, the winner! Based on circuitry from the extremely popular Orban 424A (Gated Compressor/Limiter/De-Esser: the "Studio Optimod"), the 412A offers the controls and features most demanded by audio professionals. This is a no-frills unit with all the essentials—plus no-compromise sound, performance, quality and reliability at a **very** attractive price.

Familiar ATTACK TIME, RELEASE TIME, RATIO and OUTPUT ATTENUATOR controls make operation quick and intuitive. **Both** THRESHOLD and INPUT ATTENUATOR controls are available, so you can adjust the amount of G/R exactly according to the needs of your application—the 412A can keep above-threshold output level **or** below-threshold gain constant, depending on which control you adjust. And Orban's proprietary low-distortion Class-A VCA lets you adjust the THRESHOLD control without compromising headroom or signal-to-noise ratio: They stay optimized over the control's 20dB range.

Attack and release times are program-controlled: the ATTACK TIME and RELEASE TIME controls simply scale the complex time constants faster or slower. With these controls set beyond about "12:00", the 412A produces very natural, open sound with considerable short-term dynamic range—even when the unit is adjusted for high compression ratios. As time constants are sped up, the result is increased loudness and density with a minimum of unnatural compression artifacts. So, depending on the settings of these controls, the 412A can serve as a peak limiter, a pure, gentle compressor, or a combination compressor/limiter.

The unit is utterly simple to operate, and can be used either as a "hands-off" device or as a powerful creative tool. For either use we've built-in an "automatic transmission": The threshold of limiting interacts with both the ATTACK TIME and RATIO controls to keep the peak output approximately constant regardless of control settings. (Of course, it **is** affected by the THRESHOLD and OUTPUT ATTENUATOR controls, as expected.) That way, many annoying corrective readjustments of other controls become unnecessary, making your work easier and more efficient.

This feature provides a bonus: Unlike many other units, it is impossible to severely clip the 412A's VCA unknowingly. Other potential overload conditions are indicated by two LED's: one to indicate program amplifier clipping and the other to indicate overload of the control circuitry. Between the program-controlled time constants, automatic threshold adjustments, and overload monitoring, the 412A is close to foolproof—it facilitates good, fast results even from inexperienced or overburdened operators.

Ultimately, the proof is in the **listening.** We believe that our resolutely un-trendy feedback control circuitry provides a natural sound unmatched at the 412A's modest price—and matched only by our own more sophisticated 424A. Finally, it is possible for smaller studios and production rooms, fixed installations (like churches and theaters), and travelling reinforcement systems, to get top-quality level control at an affordable price. And, because the 412A doesn't compromise basic audio quality, it is fully suited for the most demanding applications requiring its particular assortment of features.

If even more flexibility is needed, consider the 422A (single-channel) and 424A (dual-channel/stereo) Gated Compressor/Limiter/De-Essers, which combine the superb audio quality of the 412A with sophisticated gating circuitry (to prevent compression-induced noise breathing) and an effective de-esser.

Between the 412A and the 422A, Orban offers a compressor/limiter to satisfy virtually any requirement in professional audio—with unmatched quality and singular standards of documentation and customer support.

Order Guide

Model 412A Single Channel
Compressor/Limiter
RET-28A XLR-type Connector
Field Retrofit Kit
RET-29A TRS Phone Jack
Field Retrofit Kit

Model 414A Dual Channel/Stereo
Compressor/Limiter
RET-28B XLR-type Connector
Field Retrofit Kit
RET-29B TRS Phone Jack
Field Retrofit Kit

(Barrier strip connections standard, #5 screw)

Remote Control of the 275A

Remote Control Panel: An optional 19" rack-mount remote control panel provides duplication of all front-panel indicators and functions except for the SEPARATION control.

Control By Automation: The 275A has a rear-panel connector which provides optically-isolated logic inputs for automated control of noise reduction, automatic polarity correction, and synthesis functions. These inputs can accept pulsed (latching) or continuous control signals.

Pulsed signals duplicate the functions of the front-panel buttons. For example, an automated switching system could trigger the desired synthesis mode for each event, setting the 275A's mode with a short contact closure or logic pulse. The 275A would remain in that mode until given another command from the automation or front-panel controls.

Continuous control signals would be used where the desired synthesizer mode is encoded continuously, as in the vertical interval. Continuous control signals override the 275A's current mode and force it to switch to the mode specified by the control signal. When the signal ceases, the 275A returns to the mode that was active before the control signal appeared. (If a front-panel button is pressed while a continuous signal is present, the mode will not change until the signal ends—only then will the 275A return to the newly-selected mode.)

The advantage of using continuous control signals is that the 275A cannot get "hung up" in the wrong mode because the automation failed to notice that one event had ended, and that a new event requiring different 275A processing was on-line. The end of each event is automatically indicated by the end of its control signal. If the "default" mode is AUTO (as it most often will be), then there is no possibility of a catastrophic error, such as loss of audio.

The 275A provides several graceful exits from automation failure. If two or more automation control lines are active simultaneously, the unit will immediately return to AUTO recognition mode. In addition, an AUTOMATION LOCKOUT button (duplicated on the optional Remote Control Panel) allows the operator to lock out all automation signals, and to control the 275A manually until proper automation system operation is restored.

SYNTHESIZER FREQUENCY RESPONSE
IN NARROW MODE

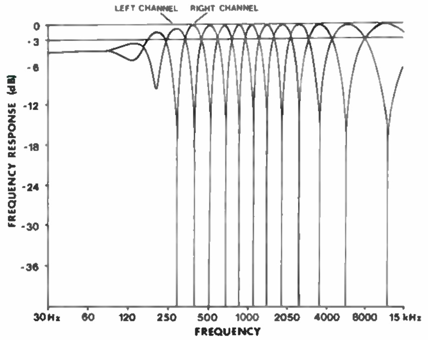


FIGURE 1

Specifications

Frequency Response
(ref mono sum):
± 1/2dB, 30-15,000 Hz (Bypass mode)
± 1dB, 30-15,000 Hz (Synthesize mode)

Total Harmonic Distortion
(+ 18dBm/600 ohms):
< 0.02%, 30-15,000 Hz (Bypass mode)
< 0.3%, 30-15,000 Hz (Synthesize mode)

Noise At Output
(30-15,000 Hz):
< – 80dBm (Bypass mode)
< – 67dBm (Synthesize mode)

Input
Impedance: > 10K ohms, balanced bridging.
Absolute overload occurs at + 26dBm.

Output
Impedance: < 100 ohms, balanced to ground.
Clipping occurs at + 26dBm into 600 ohms.

Connectors
Audio: Cinch-type 140-Y barrier strip (#5 screw).
User Control Interface: Type DB-25S jack (accepts DB-25P plug).
275A/RC Remote Control: Type DC-37S jack (accepts DC-37P plug, supplied).

Power Requirements
115-230V AC 50/60Hz, 9VA. Supplied with "U-ground" grounding-type plug to USA standards.

Mounting
Requires 1 unit (1¾", 4.5cm) of vertical space in an EIA 19" (48.3cm) rack.
Depth is 9⅝" (24.5 cm).
Optional 275A/RC remote control unit requires the same space, except depth is 2¼" (5.7cm), including supplied connector.

Shipping Weight
12 lbs. (5.4 kg)

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800041-000-02 2/87

The Orban 275A Automatic Stereo Synthesizer

An unrivaled problem-solver for Stereo TV Transmission

Features

- ☐ **Two methods of automatic mono recognition**

  "Single-Channel" detects absence of audio on one channel, "Mono/Stereo" recognizes mono in both channels.

- ☐ **Two stereo synthesis modes**

  NARROW mode effectively centers dialog, WIDE mode is more dramatic for music and effects.

- ☐ **Smooth cross-fading between true and synthesized stereo**

  No pops, clicks, or discontinuities.

- ☐ **Full mono compatibility**

  Sum of synthesized outputs is identical to mono input.

- ☐ **Patented Orban phase-shift derived comb-filter stereo synthesis technique**

  Provides synthesized stereo without addition of unnatural resonances or "flanging" colorations.

- ☐ **Automatic detection and correction of polarity-reversed ("out-of-phase") stereo inputs**

- ☐ **Noise reduction for mono material**

  Typically 10dB reduction of hiss and hum; single-ended.

- ☐ **Optically-isolated external automation control interface**

- ☐ **Optional remote control unit duplicates main unit functions**

- ☐ **Fully balanced stereo inputs and outputs**

  Can be used with +4dBm or +8dBm lines.

A Stereo Synthesizer That's Right for Stereo TV

The scarcity of true stereo program material has caused many television broadcasters to look for a device that could effectively and automatically reprocess mono material into synthesized stereo. Orban has responded to this need by developing an automatic stereo synthesizer specifically designed for in-line stereo synthesis of mono TV audio.

The **Model 275A Automatic Stereo Synthesizer** improves upon the popular, manually-operated Model 245F Stereo Synthesizer with these added features:

- ☐ Automatic mono/stereo recognition and switching with smooth cross-fade.
- ☐ A choice of *two* methods of automatic mono recognition and *two* stereo synthesis modes.
- ☐ Important stereo television "utility" features like polarity error correction and noise reduction.
- ☐ Exceptionally versatile remote control.

Manual, Automatic, or Remote Control

The 275A Stereo Synthesizer is designed to be placed permanently in the program line. Unless one of its synthesis or utility functions is selected, the 275A will pass audio transparently.

Manual controls on the front panel select whether stereo is to be synthesized from left or right channel mono, whether the "wide" (best-suited for music and effects) or "narrow" (best-suited for voice) synthesis mode is used, and whether noise reduction should be applied to the mono signal prior to stereo synthesis.

Front-panel pushbuttons also allow the operator to activate the automatic recognition and synthesis circuitry, bypass the synthesizer, lock out external automation, or route true-stereo audio through the reversed-polarity detector and corrector.

LED indicators show the functions selected, as well as the operating status of the noise reduction and polarity correction utilities.

All of these controls and indicators are duplicated on the optional, rack-mountable 275A/RC Remote Control Panel.

In addition, the 275A Stereo Synthesizer can be controlled from the station's automation system, tally, or vertical interval decoder through optically-isolated logic inputs available at a rear-panel connector.

Automatic Recognition of Mono/Stereo

It is very difficult to design a circuit which accurately distinguishes between mono and true stereo program material. What defines true mono? If the program material on the left and right channels is identical but differs slightly in level or phase due to recorder or plant errors and tolerances, is that true mono? And should a synthesizer switch-in during those segments of a stereo program when there is no stereo music or sound effects—only center-channel dialog or effects which are electronically identical to true mono?

Although electronic recognition is required, it is clear that no electronic circuit using present-day technology can perform this task perfectly—positive identification of center-channel material still requires the human ability to perceive meaning and high-level context. Nevertheless, at the request of broadcasters Orban is providing such automatic recognition, painstakingly refined and fine-tuned to make fewer errors than other center-channel recognition devices. Because some errors are still inevitable with center-channel recognition, the 275A also provides a more reliable alternative: single-channel recognition.

Single-Channel Recognition: With this approach, the station routes all mono material through the program lines on *one channel only*. If the 275A detects a mute channel, it will automatically synthesize stereo from the other, active channel. If both channels are active, the signal is considered stereo and the synthesizer is bypassed. Single-channel recognition assures reliable detection without recognition errors.

Mono/Stereo Recognition: In this technique, no special routing of program material is required. Audio present on both channels is analyzed by the 275A's recognition circuitry. Even if the program material is primarily hard-center dialog with low-level stereo music or audience noise background, the 275A will recognize it as stereo and bypass the synthesizer. If the material is electrically mono, it is routed to the synthesizer. Minor phase and level differences caused by plant tolerances are ignored. Audio present on only one channel always activates the synthesizer, as in the single-channel technique.

Regardless of which method of recognition you choose, the 275A Stereo Synthesizer cross-fades between synthesized and true stereo material smoothly and unobtrusively.

Stereo Synthesis

The patented Orban stereo synthesis technique creates a compelling pseudo-stereo effect from a mono signal by dividing the audio spectrum into several frequency bands, then directing these bands alternately to the left and right channels. It does this by passing a mono signal through a chain of phase shifters to generate an artificial L-R signal, which is then added to the mono to obtain the synthesized left channel and subtracted from the mono to obtain the synthesized right channel. The net effect is a "complementary comb filter" (see figure 1). The sum of the two synthesized channels always remains equal to the original mono, ensuring mono compatibility.

Because the audio spectrum is divided logarithmically, the undesirable harmonic reinforcement and cancellation which can result from arithmetic band-splitting is avoided. The number of bands (and therefore their individual bandwidths) determines how "dramatic" the stereo effect is. With the 275A Automatic Stereo Synthesizer, two types of remote-selectable stereo synthesis effects are available: a small number of wider bands results in a dramatic sense of stereo space on music and effects (similar to our 245F), while a larger number of narrower bands centers dialogue more accurately. A recessed SEPARATION control adjusts the amount of inter-channel difference (L-R), which determines the relative width of the stereo image.

To maximize loudness while making efficient use of the modestly-priced amplifiers in most consumer TV sets, energy below approximately 200Hz remains mono. The ear can not detect stereo separation in this region.

Noise Reduction

Older mono material often suffers from hiss and other forms of noise. The 275A can apply single-ended noise reduction to mono audio prior to stereo synthesis processing. This noise reduction combines program-controlled high-frequency filtering with broadband expansion. 10dB of noise reduction is typically achieved—without unnaturally reducing ambiance and dialog intelligibility when program levels are low.

Because the noise reduction system is single-ended, no encoding (or later decoding) of the program material is necessary. This makes the process ideally suited to noisy optical soundtracks and satellite feeds. Operation is exceptionally smooth and subtle, and "pumping" and "breathing" are entirely absent.

Noise reduction is not available for true stereo material, since the quality of most stereo material is high and the feature would not justify the additional cost.

Polarity Correction

In stereo material, it is essential that the two channels be in phase with each other. If they are not, the mono sum signal will be seriously degraded as the two channels cancel each other. And that means that the viewer with a mono set (a majority of your audience) will hear disastrously inferior audio, and in some cases no sound at all!

To ensure the mono compatibility of your stereo broadcasts, the 275A can act as a "watchdog" over your program line polarity, correcting errors when detected. The detection technique is very reliable and highly resistant to "falsing"—even when subjected to substantial high-frequency phase errors (due to misaligned heads or other mechanical problems) or when monitoring soundtracks containing out-of-phase "surround" energy.

An LED on the front (and remote control) panel lights when a polarity reversal is being corrected. The detection/correction circuit can be activated or defeated at any time.

The OPTIMOD Tradition

The 275A Automatic Stereo Synthesizer has been designed to be an integral part of Orban's OPTIMOD® Television Stereo System, along with the:

8182A Audio Processor
8185A/SG Stereo Generator
8182A/SAP Second Audio Program Generator
8182A/PRO Professional Channel Generator

The stability, reliability, and superb performance that distinguish Orban products make them the equipment of choice for the innovative stereo television broadcaster.

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Announcing Orban's New

OPTIMOD-AM Audio Processor Model 9100B

The quality and versatility of our 9100A OPTIMOD-AM *Plus* NRSC Pre-emphasis and Filtering

The new Model 9100B OPTIMOD-AM gives you all the performance and features that have made OPTIMOD-AM the choice of so many stations concerned about competing with FM. In addition, the 9100B provides the high-frequency pre-emphasis and 10kHz low-pass filtering recommended by the National Radio Systems Committee (NRSC). The 9100B increases coverage, improves source-to-source consistency, and delivers superb audio quality on both voice and music. And, incorporating the new NRSC standard allows the new OPTIMOD-AM to be even louder.

Please refer to the OPTIMOD-AM Model 9100A brochure for basic features and specifications.

The 9100B is delivered with monitor roll-off filters to apply complementary de-emphasis for off-air monitoring to accurately simulate the frequency response of a "standard" NRSC receiver. The previously optional 5kHz low-pass filter required by European broadcasters is standard on the 9100B. 5kHz, 10kHz (NRSC), and 12kHz filters can be selected by jumper or by the 9100B's day/night logic. For those whose needs are not met by the NRSC pre-emphasis standard, the 9100B can also perform high-frequency equalization according to one of three alternative pre-emphasis curves.

The 9100B OPTIMOD-AM is available in mono and stereo versions. The 9100B/1 *mono* unit is easily upgraded later to the 9100B/2 *stereo* OPTIMOD-AM with a plug-in retrofit kit. Orban also supplies retrofit kits for upgrading 9100A OPTIMOD-AM units to the equivalent of a 9100B (stereo or mono).

Why the new NRSC Standard?

Over the years, as the air waves became more crowded, interference from first and second adjacent stations became more and more of a problem. Receiver manufacturers responded by producing receivers with decreased audio bandwidth, so that the encroachment of an adjacent station's modulation extremes would not be audible as interference. This truncating of the bandwidth had the effect of diminishing the receiver's high-frequency response, but it was felt that lower fidelity would be less obnoxious than interference. As long ago as 1978, Orban proposed and implemented pre-emphasis and low-pass filtering for AM broadcast to provide brighter sound at the receiver while minimizing interference. This approach has become widely accepted. Now the NRSC has formalized a standard acceptable to all industry segments, which can, if promptly implemented, result in a vast improvement in AM radio.

AM Stereo Introduces a Pre-emphasis Dilemma

Certain AM receivers manufactured since 1984, particularly those designed for domestic AM stereo reception, have a frequency response which is substantially wider than that of the typical mono AM receiver. The frequency response was widened largely to enhance the sales potential of AM stereo by presenting a dramatic, audible improvement in fidelity in the showroom. As these new receivers became more prevalent, broadcasters had to choose whether the station's pre-emphasis would be optimized for the new AM stereo receivers or for the existing conventional receivers that form the vast majority of the market. If the choice was for conventional receivers (which implies a relatively extreme pre-emphasis), the newer receivers might sound strident or exceptionally bright. If the choice favored the newer receivers (less pre-emphasis and probably less processing), the majority of receivers would be deprived of much high-end energy and would sound both quieter and duller.

NRSC Standard Pre-emphasis and Low-pass Filtering

In response to this dilemma, the National Radio Systems Committee (NRSC) undertook the difficult task of defining a voluntary recommended pre-emphasis curve for AM radio that would be acceptable to broadcasters (who want the highest quality sound on the majority of their listeners' radios) and to receiver manufacturers (who are primarily concerned with interference from a co-channel or from first- and second-adjacent stations).

After a year of deliberation, a "modified 75-microsecond" pre-emphasis/de-emphasis standard was approved. This provides a moderate amount of improvement for existing narrowband radios, while optimizing the sound of wideband radios. Most importantly, it generates substantially less first-adjacent interference than do steeper pre-emphasis curves.

The second part of the NRSC standard calls for a sharp upper limit of 10kHz for the audio presented to the transmitter. This essentially eliminates interference to second and higher adjacencies. While some have protested that this is inadequate and that 15kHz audio should be permitted, the unfortunate fact is that interference-free 15kHz audio could only be achieved by a complete re-allocation of the AM band! The *practical* effect of widespread implementation of the 10kHz standard is that 10kHz radios will then be feasible, and the bandwidth perceived by the average consumer (now limited by the receiver to 3kHz, typically) will be dramatically improved. On much mass-market consumer equipment, it will be difficult to tell AM from FM.

The radio manufacturers participating in the NRSC stated emphatically that reduction in interference *must be demonstrated by broadcasters* before receiver manufacturers would be willing to release true wideband (10kHz audio bandwidth) receivers to the mass market. This is rational – the receiver manufacturers can lose millions of dollars if they produce receivers that are rejected as noisy or interference-prone by consumers. In contrast, broadcasters can easily change pre-emphasis and filtering with very little expense.

Therefore, although this standard is voluntary, we *strongly recommend* conformance to it. We are convinced that use of this more modest pre-emphasis and sharp 10kHz filtering by broadcasters is the only factor that will eventually induce the receiver manufacturers to build and mass-market the high-fidelity, wideband radios which would allow AM stations to compete with FM in audio quality. The commitment to do so was strongly expressed by the receiver manufacturers involved in the NRSC's deliberations.

Orban is doing its part — we have designed the new 9100B to conform to the full NRSC standard, and we can supply low-cost conversion kits for every OPTIMOD-AM ever made. The ball is now clearly in the broadcasters' court.

Estimated first delivery: May, 1987

Price: $4395 for Model 9100B/1 (mono)
$5995 for Model 9100B/2 (stereo)

800050-000-01 3/87

Orban Associates Inc. 645 Bryant Street San Francisco, CA 94107
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Upgrade Kits for OPTIMOD-AM®

NRSC Voluntary Standards for AM Broadcast

75µs (modified) Pre-emphasis   10kHz Low-pass Filtering

In the few short years since its introduction, OPTIMOD-AM Model 9100A has grown to be one of the most-often used tools for improving AM audio.

Now there is new promise for AM improvement. Over a year ago, the National Radio Systems Committee (NRSC) brought broadcasters, equipment manufacturers, and receiver manufacturers together to talk about a voluntary national transmission standard that would make wideband high-fidelity AM radios practical. Today, after hundreds of hours of discussion and study, the standard finally exists. It consists of a "modified 75µs" pre-emphasis curve which brightens up the sound on older radios while minimizing first-adjacent interference, plus a very sharp cut-off 10kHz low-pass filter which dramatically reduces splatter into the second-adjacencies.

The receiver manufacturers participating in the NRSC stated that such reduction in interference *must be demonstrated by broadcasters* before receiver manufacturers would be willing to release true wideband (10kHz audio bandwidth) receivers to the mass market. This is rational — the receiver manufacturers can lose millions of dollars if they produce receivers that are rejected by consumers as noisy or interference-prone. In contrast, broadcasters can change their pre-emphasis and low-pass filtering at very little expense.

Orban was the first to propose AM pre-emphasis and low-pass filtering, and was heavily involved in the Committee's work and research. We strongly endorse the new NRSC standards. They're good engineering *and* good business, and we urge all AM broadcasters to comply as soon as possible.

To underscore our commitment to the new standards, we designed the following modules and cards for Model 9100A OPTIMOD-AM to add the NRSC-standard pre-emphasis and filtering.

Components of Upgrade Kits:

BLUE NRSC 75µs Pre-emphasis Equalization Module

When installed in the 9100A, the BLUE Module provides pre-emphasis on the curve recommended by the NRSC when the 9100A's HF EQ control is turned fully clockwise. Two such modules are required for stereo. For OPTIMOD-AM Models 9100A/1 and 9100A/2 with serial numbers below 700000, the RET-033 Retrofit Kit (listed below) must also be installed.

#1F10 NRSC Mono Low-pass Filter Card

The #1F10 Card offers three individually selectable functions that can be jumpered to follow day/night switching in any combination: (1) NRSC-standard 10kHz low-pass filter for all North American stations, (2) 5kHz low-pass filter for international use (other filter frequencies are available on special order), and (3) bypass (12kHz overall processor bandwidth).

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Revision 09, Effective 1 April 1987
Supersedes Revision 08, 1 April 1986
Changes: NEW OAM models and Retrofit Kits to incorporate NRSC-standard pre-emphasis and low-pass filter; price increase. 9100A/1 superseded by 9100B/1; 9100A/2 by 9100B/2; 9100A/2C by 9100B/2; ACC-005 by ACC-023; RET-016 by RET-043A; RET-031 by RET-041; RET-032 by RET-042; RET-038 by RET-043A.

ORDERING GUIDE & SUGGESTED LIST PRICES

OPTIMOD-AM BROADCAST PRODUCTS

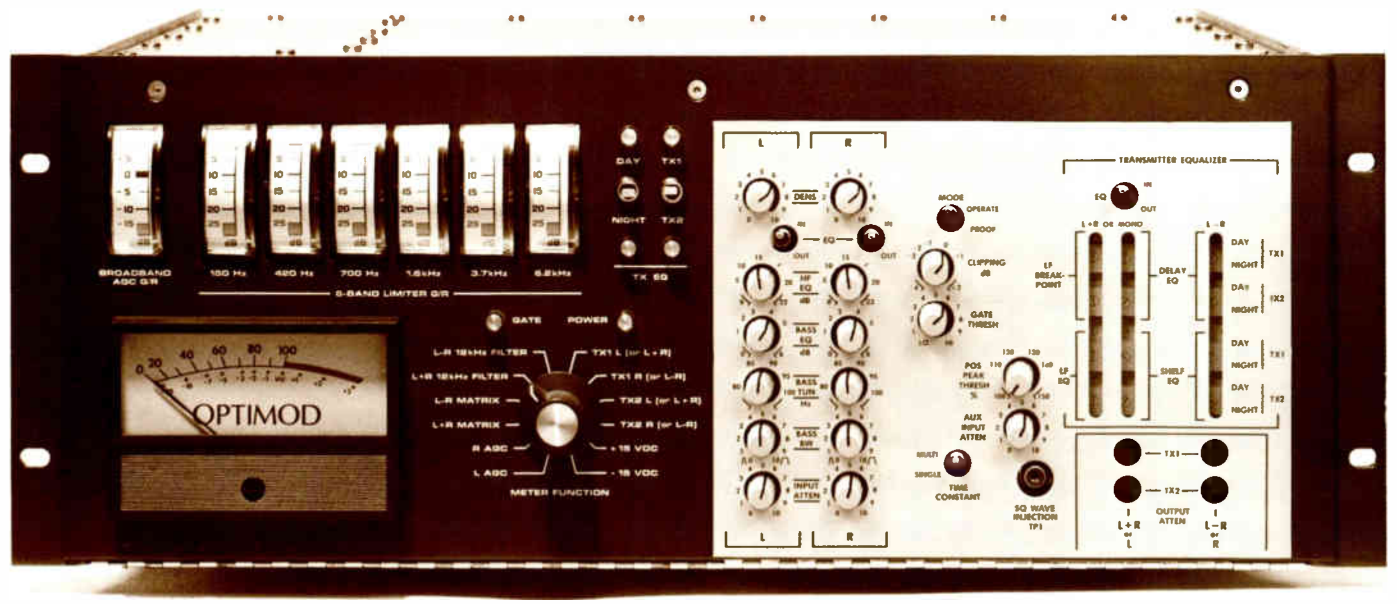
Model	Description	USA List Price
	OPTIMOD-AM	
9100B/1	**OPTIMOD-AM AUDIO PROCESSING SYSTEM (MONO)**	$4,395.00
	Complete audio processing for AM broadcast. Includes broadband AGC, NRSC-standard and alternative pre-emphasis, six-band limiter with distortion-cancelled clipper, switchable NRSC 10kHz filter, jumperable 5kHz low-pass filter, transmitter equalizer for two transmitters day/night. One ACC-023 NRSC Monitor Rolloff Filter supplied. Field convertible to stereo. 115V/230V, 50-60Hz.	
9100B/2	**OPTIMOD-AM AUDIO PROCESSING SYSTEM (STEREO for C-QUAM or Kahn)**	$5,995.00
	As 9100B/1 above, equipped for stereo operation. Uses sum and difference control of processing to assure maximum loudness on mono receivers. Switchable features include L and R 75% negative peak limiter as recommended by Motorola for C-QUAM, adjustable stereo enhancer that increases L-R, 200Hz high-pass filter for telemetry or LF SCA. Two ACC-023 NRSC Monitor Rolloff Filters supplied. 115V/230V, 50-60Hz.	
	AM ACCESSORIES	
ACC-023	**NRSC Monitor Rolloff Filter (one per channel)**	$50.00
	Approximates typical receiver rolloff when monitoring from modulation monitors and wideband receivers. Includes rolloff to the NRSC standard 75us de-emphasis. Included in all 9100B/1, 9100B/2 and several Stereo Upgrade Kits.	

See reverse side for UPGRADE KITS and RETROFIT KITS

Prices are FOB ORBAN ASSOCIATES INC., SAN FRANCISCO, and are subject to change without notice. Orban Broadcast Products are sold through authorized Orban Broadcast Dealers worldwide. For names of Dealers near you, or for more information, call or Telex Orban.

Orban Associates Inc.
645 Bryant Street, San Francisco, CA 94107 USA
Telephone (415) 957-1067 or (800) 227-4498 Telex 17-1480

Model	Description	USA List Price
	UPGRADE KITS FOR OPTIMOD-AM MODEL 9100B	
RET-043B	**Stereo Upgrade Retrofit Kit** Upgrades 9100B/1 to 9100B/2 (STEREO for C-QUAM or Kahn). Includes Cards #1S10 to replace #1F10, Cards #5, #9, #10, and one additional ACC-023 NRSC Monitor Rolloff Filter.	**$1,895.00**
	UPGRADE KITS FOR OPTIMOD-AM MODEL 9100A	
RET-033	**Alternate Pre-emphasis Module Retrofit Kit** For 9100A/1 and 9100A/2 prior to S/N 700000. (Supplied standard on later units.) Provides socket for field-selectable alternative pre-emphasis curves. Includes one Transition PCB Assembly, one each GREEN, YELLOW, RED Pre-emphasis Module. Two RET-033 kits required for stereo.	**$50.00**
RET-041	**NRSC Pre-emphasis/Filter Retrofit Kit (MONO)** For 9100A/1. Provides NRSC-standard pre-emphasis and 10kHz low-pass filter, and jumperable 5kHz low-pass filter (other filter frequencies available on special order). Filters can be preset or switched to follow day/night switching. Includes one #1F10 NRSC Mono Filter Card, one NRSC Blue Pre-emphasis Module, one set NRSC monitor de-emphasis resistor and capacitor. (RET-033 must first be installed in units prior to S/N 700000.)	**$295.00**
RET-042	**NRSC Pre-emphasis/Filter Retrofit Kit (STEREO)** For 9100A/2 and 9100A/2C. Provides switchable NRSC-standard pre-emphasis and 10kHz low-pass filter, jumperable 5kHz low-pass filter (other filter frequencies available on special order), L and R 75% negative peak limiter as recommended by Motorola for C-QUAM, adjustable stereo enhancer that increases L-R, 200Hz high-pass filter for telemetry or LF SCA. Filters can be preset or switched to follow day/night switching. Includes one #1S10 NRSC Stereo Filter Card, two NRSC Blue Pre-emphasis Modules, two sets NRSC monitor de-emphasis resistor and capacitor. (Two RET-033 must first be installed in units prior to S/N 700000.) $225 trade-in for return of a #1-S Card previously installed -- ask your Orban Broadcast Products Dealer for details.	**$495.00**
RET-043A	**Stereo Upgrade Retrofit Kit** Upgrades 9100A/1 to 9100B/2 (STEREO for C-QUAM or Kahn). Adds stereo NRSC-standard pre-emphasis and low-pass filter. Includes Cards #1S10, #5, #9, #10, two NRSC Blue Pre-emphasis Modules, two ACC-023 NRSC Monitor Rolloff Filters.	**$1,950.00**
	UPGRADE KITS FOR OPTIMOD-AM MODEL 9000A	
RET-044	**NRSC Pre-emphasis/Filter Retrofit Kit** For 9000A and 9000A/1. Provides NRSC-standard pre-emphasis and 10kHz low-pass filter, and jumperable 5kHz low-pass filter (other filter frequencies available on special order). Includes one #1F10 NRSC Mono Filter Card, modification parts to install filter card and pre-emphasis, one set NRSC monitor de-emphasis resistor and capacitor. (Requires considerable rework of existing cards and backplane.)	**$395.00**



SUBPANEL SETUP CONTROLS
(ACCESS DOOR REMOVED)

(continued from inside)

The 9100A incorporates a phase scrambler early in the system to make peaks as symmetrical as possible. Operated symmetrically, the 9100A will produce an extremely loud and silky-clean sound, free from the midrange "grit" so often associated with the sound of soft-clipping asymmetrical AM processors. You really *can* hear the absence of even-order IM in the midrange! However, for those broadcasters desiring the loudest possible sound, a POSITIVE PEAK LEVEL control permits you to adjust asymmetry to beyond +125% modulation. The choice is yours, and will depend on format, target audience, and other programming considerations.

Because of the phase scrambler, any natural asymmetry in the input material is eliminated and most efficient use is made of the processing. For this reason, no automatic polarity switching is included (or desirable) in the 9100A. Any asymmetry at the output is artificial and is produced by the processing itself as controlled by the user.

Summary

The 9100A is an exciting development in AM audio processing. For the first time, it permits high loudness to be achieved along with the openness and freedom from processing artifacts heretofore only associated with such FM processors as our OPTIMOD-FM Model 8100A.

While the 9100A is ideally suited for *any* stereo system, the mono station will benefit fully from its use. FM's edge is not so much stereo as it is *quality*. Many now feel that AM has been *damaging itself* by being strident, busy, and overprocessed. With the 9100A, programmers and engineers now have a friendly, well-honed tool to create a sound which complements their creative programming: a sound which feels much like FM in its openness, depth, and definition. Many of the hundreds of stations already using 9100A processing have found that ratings have moved in the right direction: Fatigue is reduced, so more people listen longer. This is particularly important for those stations—whether talk or music—orienting their programming toward adult demographics.

For many AM broadcasters—particularly those with music formats—the sound of the 9100A may be the key to recapturing audience and ratings.

Order Items

[Stereo generators for the various AM stereo systems are not sold by Orban and must be obtained separately.]

9100A/1	Mono Unit (Field-convertible later to stereo using plug-in cards [RET-16].)
9100A/2	Stereo unit (Can be readily used for mono.)
RET-32	Stereo Compatibility Card #1-S (see text)
RET-31	Mono Lowpass Filter Card #1-F (see text)

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Specifications

It is impossible to define the listening quality of even the simplest limiter or compressor on the basis of the usual specifications because such specifications cannot adequately describe the crucial *dynamic* processes which occur under program conditions. These dynamic processes are evaluated at the factory and controlled to close tolerances. The measurements require special test fixtures, cannot be readily duplicated by measurements with standard test equipment in the field, and cannot be described in familiar terms.

Certain specifications are therefore presented here to satisfy the engineer that they are reasonable, to help plan the installation, and to help make certain comparisons with other processing equipment.

The 9100A is the result of rigorous computer-aided design, high-precision components, careful testing, and extensive burn-in. In contrast to some competitive equipment, its PROOF/OPERATE switch defeats clipping and gain reduction, but *does not bypass any circuitry*. (The receiver equalizer can be defeated with a separate switch to perform performance measurements under "flat" conditions.) Thus its specifications, even in PROOF mode, truly reflect the accuracy and transparency of its signal path. We have nothing to hide; we are proud of the 9100A's performance.

INPUT

Impedance: Greater than 10K ohms, electronically balanced by means of true instrumentation amplifier. RF suppressed.
Sensitivity: Normal operation may be achieved with nominal line levels of −30 dBm or greater. Input sensitivity is controlled by means of INPUT ATTENUATOR control, and also by means of a bypassable 20dB pad before the input amplifier.

INPUT CONDITIONING FILTER

Highpass: −0.5dB @ 50 Hz with rolloff 18 dB/octave below that frequency. Includes deep 25Hz notch for automation cuetones, and to protect the C-QUAM pilot tone.
Lowpass: Rolls off frequencies above 12kHz at a rate exceeding 24dB/octave.
Allpass: Phase scrambler makes peaks more symmetrical to best utilize the capabilities of the processing.

BROADBAND COMPRESSOR

Range Of Compression: 25dB.
Compression Ratio: greater than 10:1.
Time Constant Mode: SINGLE/MULTI, switch-selectable.
Attack Time: approximately 200ms (SINGLE); program-controlled (MULTI).
Release Time: approximately 3dB/second (SINGLE), program-controlled (MULTI).
Total Harmonic Distortion: does not exceed 0.05% at any degree of gain reduction, 50-12,000 Hz.
Noise: greater than 85dB below output clipping level.
Gating: gain will drift slowly to 10dB G/R if input level drops below a user-adjusted threshold.
Variable-Gain Element: proprietary class-A VCA

PROGRAM EQUALIZER

Bass: "Quasi-Parametric" second-order peak boost equalizer. "Q" variable from 0.3 to 1.4. TUNING variable from 70 to 110Hz. EQ variable from 0 to +6dB.

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High Frequency: Proprietary third-order shelving equalizer (patent pending) matches inverse of "average" (see text) receiver rolloff to a high frequency limit, adjustable from no equalization at all, through any frequency up to 6kHz. Three different curve families are selectable by means of plug-in modules.

SIX-BAND LIMITER

Filters: 150Hz lowpass; 420Hz bandpass; 700Hz bandpass; 1.6kHz bandpass; 3.7kHz bandpass; 6.2kHz highpass.
Filter Selectivity: 18dB/octave.
Filter Topology: parallel.
Filter Combination: Static: outputs of all filters combine to yield static frequency response ±0.5dB throughout the range of 50-12,000Hz.
Dynamic: Phase interaction between filters under program conditions will not cause audible dips in the frequency response.
Limiters: Range: 25dB (Bands 1-5); 30dB (Band 6).
Attack Time: program controlled; adjusted according to band frequency.
Release Time: program controlled; adjusted according to band frequency.
Total Harmonic Distortion (each limiter): does not exceed 0.1% for any frequency in each limiter's passband with any degree of gain reduction, provided signal is below multiband clipper threshold.
Distortion Cancellation: all clipper-induced distortion in upper four bands cancelled better than 30dB below 1.8kHz. Additional distortion reduction provided as function of frequency in each band.
Noise (each limiter): better than 85dB below VCA output clipping level.
Variable-Gain Elements: proprietary class-A VCA's.

OUTPUT FILTER

Filter Characteristics: 12kHz 5th order elliptical is standard. System guaranteed to meet all requirements of FCC 73.40.a.12 regarding occupied bandwidth for arbitrary adjustments of processing controls and arbitrary program material, provided that the transmitter does not add significant high-frequency harmonic distortion to its spectrum.
Optional Filters: Optional plug-in Card #1-S (for stereo or mono applications) contains two phase-corrected 5kHz 30dB/octave filters for L+R and L−R processing). These can be strapped in or out as desired, or configured to follow the DAY/NIGHT logic. Card #1-S also contains a strappable 200Hz 30dB/octave L−R highpass filter and phase-matching L+R allpass filter to protect L−R low frequency SCA or telemetry in stereo applications. Optional Card #1-F (for mono applications *only*) contains a 5kHz lowpass filter with logic and electronic switching equivalent to 5kHz lowpass filters in the stereo #1-S card.

TRANSMITTER EQUALIZER

Low Frequency Tilt Equalizer: Proprietary phase/magnitude compensation introduces adjustable positive-slope tilt to the output waveform to cancel normal negative-slope tilt in older-technology transmitters. Independent control of very-low-frequency compensation available to avoid saturation and non-linear effects in transmitters with limited frequency power handling capacity.
High Frequency Shelving Equalizer: Adjustable breakpoint shelving equalizer creates controlled undershoots in high frequency transient waveforms to prevent RF envelope overshoot due to excessive "Q's" in transmitters, phasors, and/or antenna systems, or due to poor transient response in audio or modulator stages.
High Frequency Delay Equalizer: Introduces added time delay selectively into the spectrum to compensate for non-linear group delay in the transmitter/antenna system.
Controls: Four separate sets of controls are provided which can be independently adjusted for DAY/TX1, DAY/TX2, NIGHT/TX1, and NIGHT/TX2, and can be remotely switched by momentary application of 6-24V AC/DC between the appropriate terminals on the rear-panel barrier strip. DAY/NIGHT and TX1/TX2 status is indicated by pairs of LED's on the front panel.

LINE DRIVER

Output Impedance: 290 ohms, electronically balanced to ground. RF suppressed by means of third-order non-overshooting EMI filter.
Output Level: will drive greater than +20 dBm into 600 ohms.
Configuration (mono): Outputs for two transmitters, each with independent TX EQ and 18-turn screwdriver-adjust OUTPUT ATTENUATOR controls.
Configuration (stereo): Outputs for two transmitters, each output with independent TX EQ and 18-turn screwdriver-adjust OUTPUT ATTENUATOR control. Outputs can be strapped for L&R, or L+R and L−R depending on the needs of the subsequent stereo generator.

COMPLETE SYSTEM (PROOF MODE)

Frequency Response: better than ±1.0dB, 50-7500Hz (optional 5kHz filters defeated).
Total Harmonic Distortion: less than 0.2% at 100% modulation, 50-7500Hz.
RMS Noise: better than 75dB below 100% modulation, 30-20,000Hz.
Separation (9100A/2): Better than 20dB, 50-7500Hz; 30dB typical

SUPPLEMENTARY STEREO SPECIFICATIONS

Dynamic Separation (Pink Noise; OPERATE Mode): 30dB typical.
Optional Card #1-S: Contains subsystem to protect C-QUAM receivers by limiting left or right channel negative peak modulation to −75%, where −100% is full negative envelope modulation. Control occurs by means of asymmetrical clipping in L and R channels. Clipper overdrive under conditions of high separation is prevented by dynamic gain reduction in L−R channel prior to L and R clipping (variable blend). L−R gain reduction is constrained to 6dB or less, as required by program material. Mono receivers are unaffected, retaining full loudness. STEREO ENHANCE control on card permits increasing gain of L−R channel up to 6dB. Usable with all stereo systems. Card also contains lowpass and highpass filter subsystems. (See **Optional Filters** above.)

PHYSICAL

Dimensions: 19"W × 7"H × 12.5" D. (4 EIA rack units); 48.3cm × 17.8cm × 31.8cm).
Operating Temperature: 0-50°C (32-122°F).
EMI Environment: Circuitry shielded against EMI from 500kHz-1GHz.
Access: Circuitry (except for power supply regulator) on plug-in cards. All circuitry and user setup adjustments available from front panel without removing unit from rack. Control access door is fitted with a lock to prevent unauthorized adjustments.
Power Requirements: 115/230V AC ± 15%, 50/60 Hz.
Maximum AC Leakage To Chassis: 0.25mA @ 115 VAC; 0.5mA @ 230VAC.

WARRANTY

One year, parts and labor. Subject to limitations set forth in our Standard Warranty Agreement. Factory assistance and service available throughout the life of the product.

9100A OPTIMOD®-AM

STEREO OR MONO

OPTIMOD-AM 9100A: IT BRINGS OUT YOUR BEST.

An integrated audio processing system for AM stereo or mono, including compressor, program equalizer, multi-band limiter, clipper, and transmitter equalizer.

Highlights

- ☐ Supplies very high average modulation (loudness), exceptional fidelity, uncanny naturalness, and freedom from processing artifacts to yield an FM-like sound from *typical* auto, portable, and table radios.
- ☐ Designed for universal application in domestic or international LW, MW, and SW services; easily adaptable to relevant government standards.
- ☐ Compensates for receiver high frequency rolloff with statistically-derived, adjustable equalizer to extend perceived bandwidth at the receiver without introducing mid-range response anomalies or receiver tuning difficulties.
- ☐ Six-band limiter with distributed, distortion-cancelled multiband clipper provides dramatic increase in RMS modulation levels without pumping.
- ☐ Consistent output level and equalization texture over a 25dB input level range.
- ☐ Versatile and simple setup controls let you quickly arrive at the sound you really want.
- ☐ Improved transmitter equalizer corrects tilt and ringing in older transmitters and antenna systems for maximum modulation.
- ☐ Outputs for *two* transmitters with independent output level control and full remote control switching for TX1/TX2 and DAY/NIGHT status: i.e., *four* sets of TX EQ adjustments available, all remote-selectable.
- ☐ Sum-and-difference stereo or mono versions available; mono unit fully ready for stereo processing by simply plugging in additional circuit cards.
- ☐ Stereo version with optional #1-S Card fully approved by Motorola for use in C-QUAM installations.
- ☐ Orban-quality construction, documentation, support, and service.

FM-Like Quality— For AM Survival

Since its inception in 1982, OPTIMOD-AM Model 9100A has firmly established itself as the processor of choice for those AM stations demanding high-quality, *natural* sound free from the pumpiness, grittiness, and "honky" midrange colorations that some broadcasters used to think were necessary to achieve competitive AM loudness. Unfortunately, some processor manufacturers have not yet abandoned the old approach—even as the audience stampedes to FM.

To be sure, OPTIMOD-AM *is* a loud processor—but loudness is not the only point. In an age in which AM audience share is steadily being lost to FM, *quality* is also vital. And FM-like *quality* is what OPTIMOD-AM delivers—at the loudspeakers of typical auto, portable, and table radios—with solid, competitive loudness.

There is no one technical feature of OPTIMOD-AM which accounts for its ability to supply high quality and high loudness simultaneously. OPTIMOD-AM is a very sophisticated *system*, designed *from the ground up for stereo*, and employing many subsystems working synergistically. Many of these subsystems are exclusive to Orban.

Two of the more important are our adjustable receiver equalizer which compensates for receiver high-frequency rolloff to achieve a bright, FM-like sound, and our patented multiband distortion-cancelled clipper which eliminates any need to employ fast wideband gain reduction anywhere in the system. Accordingly, we use only *two* cascaded stages of gain reduction: a slow gain-riding AGC which acts as a gentle "hand on the pot", and a six-band limiter with steep-slope crossovers which prevents unnatural modulation of one part of the frequency spectrum by energy in another part, and which assures consistent texture and tonal balance from one program source to the next.

Many consultants now believe that such consistency is one of the keys to a polished, professional, audience-building sound. Our six-band limiter is so effective in providing this consistency that laborious re-equalization and processing in the production studio are almost never required. In addition, the six-band limiter produces a crisp, authoritative, and highly intelligible sound when used in live-voice formats (such as News and Talk). It is uniquely valuable in pulling sources with unpredictable quality like telephone calls and actualities "out of the mud" automatically.

In contrast to certain competitive processors which, incredibly, cascade as many as *six* non-cohesive stages of AGC, the payoff from our elegantly simple design is a dramatically open, effortless, multiband sound with literally *no* audible processing on virtually any radio likely to be in the hands of your audience. It is a sound which is "FM-like" not only in terms of frequency response, but also in terms of "punch", "depth", "openness", and "definition". And a sound which, on a true RMS meter, averages about 3dB higher than our previous AM processor for the same peak modulation. In short, a *loud* yet astonishingly natural sound which, we think, stands the best chance yet of winning back an audience becoming more and more attracted to FM.

The AM Stereo Processor That's Ideal For Mono Receivers

The 9100A is available as a stereo processor or as a stereo-convertible mono processor. Stereo conversion is achieved simply by plugging in circuit cards—no extra-cost "accessory chassis" is required. [The stereo generator, however, must be obtained from others.]

Processing occurs in the "sum-and-difference" mode, which is most appropriate for AM stereo because the AM modulation component represents the sum (L+R) of the channels, assuring compatibility with mono receivers. Internal straps determine if the output is to be in L+R and L−R mode, or in L and R mode, yielding complete versatility in matching it to the stereo generator.

An optional plug-in stereo compatibility card (Card #1-S) contains several circuits which permit you to tailor the 9100A ideally to *any* stereo system. These circuits can be strapped IN or OUT in any combination.

A single-channel limiter circuit protects C-QUAM stereo receivers by preventing single-channel modulation from exceeding −75% negative *without* adversely affecting mono receivers. This circuit also offers a "stereo enhance" control (usable with *any* stereo transmission system) which dynamically increases stereo (L−R) information up to 6dB, significantly increasing ambience and stereo loudness. The 9100A has been evaluated in Motorola's laboratory and has been fully approved for use in C-QUAM installations.

The second circuit can be used to protect low-frequency L−R SCA or telemetry. It contains an L−R 200Hz highpass filter and phase-matching L+R allpass filter.

The standard bandwidth of the 9100A is limited to 12kHz by means of highly selective filters, enabling it to comply with the occupied bandwidth requirements of FCC 73.40.a.12 with arbitrary program material and processing adjustments. Card #1-S contains a third defeatable circuit: dual 30dB/octave 5kHz lowpass filters in the L+R and L−R channels. These can be inserted in the signal path at all times, or strapped to follow the DAY/NIGHT logic. This enables broadcasters to limit mono or stereo bandwidth to 5kHz at night (to control interference to other stations), or to operate at 5kHz at all times to meet EBU or other international specifications. (If 5kHz bandwidth limitation is required in a *mono* 9100A/1, order the less-expensive Card #1-F which performs the function of the third circuit, but in *mono only*.)

The 9100A: Designed For The Real World

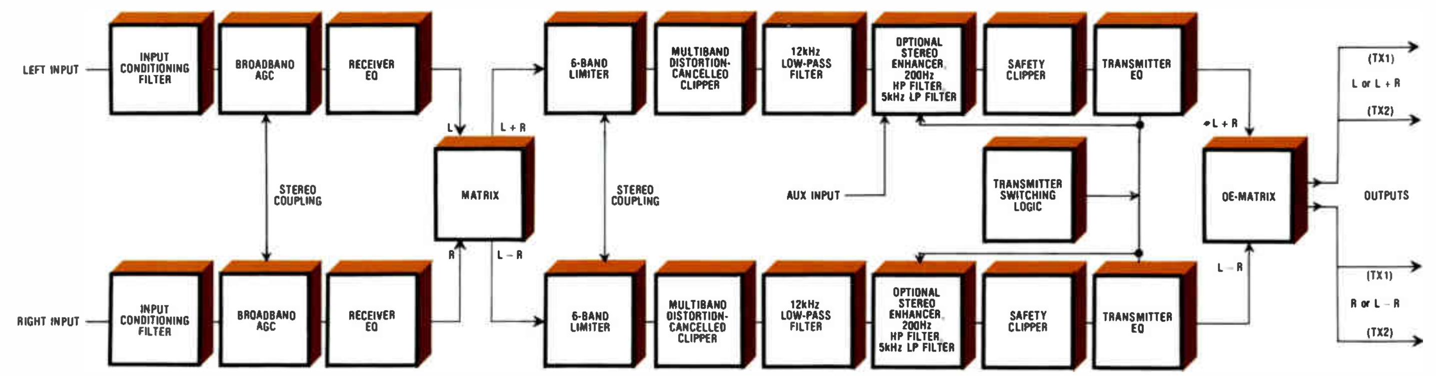
Operator Gain-Riding: A classic problem of multiband compressors is excessive sensitivity to input level variations. Incorrect operator gain riding can change frequency balances and equalization textures in a disturbing way.

Because of this, the 9100A has an AGC amplifier ahead of its six-band limiter to slowly and subtly gain-ride over a 25dB range. Accordingly, despite its being a wideband device, it does not create nasty wideband processing artifacts. And, because manual gain-riding is far less critical, the operator has more time to devote to the creative aspects of the programming—even in a high-pressure combo operation.

Receiver Equalization: To determine the optimum equalization curve that would best complement most receivers, a large number of real world AM radios were measured. The data was processed using statistical averaging techniques to assure that the average was meaningful. The resulting curve effectively flattens average receivers to 6kHz (at −3dB). For stations who wish to use less equalization because of receiver tuning considerations or transmitter limitations, the EQ can be reduced as desired. At any EQ setting, the resultant receiver output is free of mid-range peaking and coloration—an impossible outcome with conventional triband processors if such equalization is attempted.

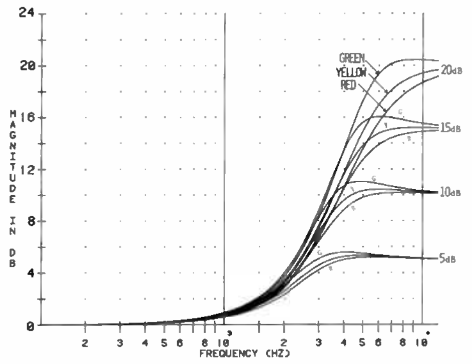
Because this preemphasis would be excessive if extended all the way to 12kHz, it *must* shelve-off at high frequencies. *How* it is to shelve-off cannot be predicted from engineering measurements alone: your subjective judgement is required. You set the *frequency* at which the curve shelves-off by using the HF EQ control to determine how far the effective bandwidth of receivers is extended by the pre-emphasis.

To provide further opportunity to "tune" the processing to the receivers prevalent among your target audience, three plug-in modules are supplied. Each provides a different *curve family* for the HF EQ control.



SYSTEM BLOCK DIAGRAM
(STEREO CONFIGURATION)

Changing modules varies the *curve shape* just before the shelf. The green module (our "standard" equalizer) provides a slight presence peak before the shelf. It is ideal for stations optimizing their sound for typical narrowband mono radios, but it may cause stridency on some wideband stereo radios. The red module provides less upper midrange energy, creating a smoother sound on the new wideband stereo radios at the expense of presence on narrowband radios. The yellow module splits the difference.



TYPICAL EQ CURVE SHAPES

There are no superfluous equalization controls which might be claimed to add "versatility", but which are in fact easy to misadjust. *All* of our curve families are scientifically correct in the sense that they match the "average" receiver in the midrange and therefore cannot create a "honky", fatiguing lower-midrange boost.

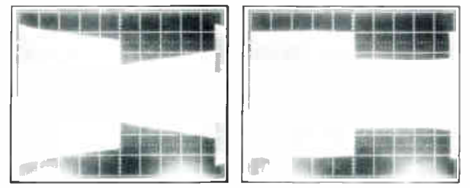
To protect your investment, our modules are low-cost plug-in resistor arrays, so we can easily provide economical, technically-correct updates if average receiver frequency response standards change as AM stereo technology advances over the years.

Because receiver bass performance is so inconsistent from model to model, it is impossible for us to recommend a scientifically-valid "universal" bass equalization curve. Instead, we include a versatile *parametric* bass equalizer which can tune bass response to your format and target audience.

Finally, to keep D.J.'s happy, we provide a passive monitor rolloff filter with an HF ROLLOFF control to match it to the processor HF EQ setting you've chosen. This filter can be inserted between your mod monitor output and your monitor amplifier input to produce a big, high-fidelity sound for local off-the-air monitoring even on large studio monitor speakers.

Transmitter Equalization: Not everyone is fortunate enough to operate a state-of-the-art transmitter with its negligible tilt and overshoot. Older plate-modulated transmitters can have enough tilt and/or overshoot to substantially compromise the accuracy with which they can reproduce a highly-processed signal like the output of the 9100A. And even state-of-the-art transmitters can ring into high "Q" antenna systems. Often, average levels must be reduced to accommodate the peak level increases introduced by such inaccuracies.

The 9100A is equipped with a transmitter equalizer to "tune out" tilt and overshoot. An important refinement allows you to control the amount of very-low-frequency correction introduced, permitting you to match the equalization to the transmitter as accurately as possible while simultaneously avoiding saturation of the modulation transformer (or other circuitry) in transmitters which cannot take the full correction.



WITHOUT TILT CORRECTION — WITH TILT CORRECTION

LOW FREQUENCY SQUARE WAVE RESPONSE

Protecting STL's: Unlike our current OPTIMOD-FM and -TV, the 9100A system cannot be split into separate studio and transmitter sections to provide compression before the STL. To accommodate those who need overload protection for their STL's, we recommend the use of an Orban 422A (mono) or 424A (stereo) compressor/limiter/de-esser at the studio side of the STL.

THE 424A

These economical, high-quality units are well-matched to the 9100A, costing no more than a "9100A Studio Accessory Chassis" otherwise would.

Auxiliary Input: A separate input is provided after principal processing but before the safety clipper and transmitter equalizer. Located in the L+R (or mono) channel, it provides a convenient port for injecting EBS tones, subsonic telemetry, and the like. Stations which desire a separate voice-processing audio chain can mix its output into the Auxiliary Input and use the 9100A processing for music only.

Construction And Maintenance: The 9100A is packaged on plug-in cards. This packaging technique protects your investment by facilitating maintenance and permitting low-cost updates as receiver and AM stereo technology advances. The cards plug into a rugged chassis with effective, field-proven RFI shielding, making the system operable in almost any EMI environment without difficulty.

Orban customer service is available toll-free to answer questions and to assure a successful installation. Those wishing to do in-house maintenance of the 9100A system will be greatly aided by an outstandingly complete operation and maintenance manual which contains detailed circuit descriptions, standard curves, and other helpful data.

Asymmetry And Polarity Followers

Much careful listening has convinced us that symmetrical *is* cleaner. Any processor which uses aggressive amounts of clipping for peak limiting will produce only odd-order harmonic and IM products when clipping symmetrically. Asymmetrical clipping produces *both* odd- and even-order products, sounding somewhat brighter, significantly dirtier, and only very slightly louder.

>

PRELIMINARY (Subject to Change)

orban

The New Orban 642B Parametric Equalizer

The Next Generation of Parametric Excellence

- 4-band, dual-channel parametric equalizer. Each section offers overlapping tuning with a 25:1 frequency range; +16dB boost/−40dB cut in each band.
- Bandwidth variable from about 5 to 1/4 octaves: ("Q": 0.29 to 5.0).
- "Constant-Q" design enables use of equalizer as a true infinite-depth notch filter; vernier frequency control on each band facilitates precise tuning of notches.
- Tunable 18dB/octave high-pass filter and 12dB/octave "Automatic Sliding Besselworth"™ low-pass filter provide maximum flexibility while preserving musicality.
- Front-panel **Cascade** switch permits use as either a two-channel 4-band, or one-channel 8-band equalizer.
- In/out switches on each band and on each channel simplify comparison of EQ settings with flat.
- 12dB make-up gain is available.
- Overload indicator warns of overload anywhere in equalizer.
- Noise and distortion specs significantly better than 16-bit digital.
- Active balanced inputs and outputs with optional output transformer.
- Main signal path is free of coupling capacitors.

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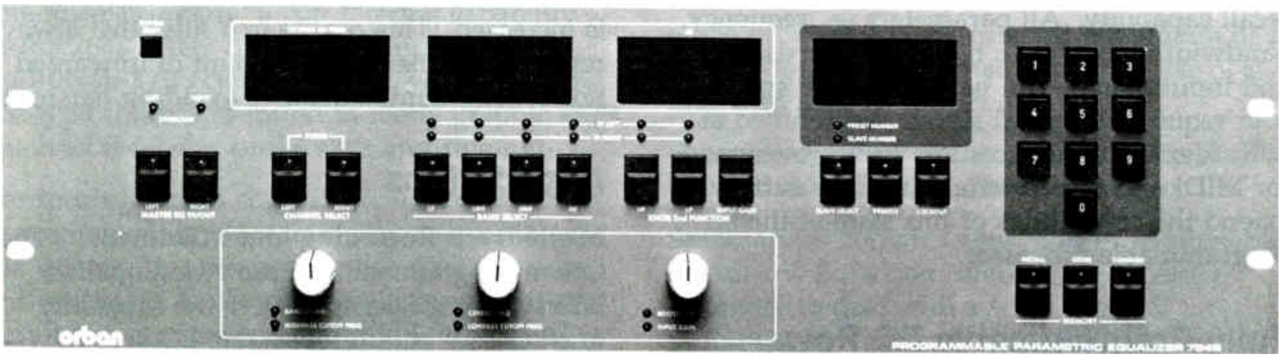
800055-000-01 10/87

PRELIMINARY (Subject to Change)

The New Orban 764B Programmable Parametric Equalizer

A major advancement in EQ technology



The Orban 764B Programmable Parametric Equalizer represents the most powerful equalization tool available for audio professionals — one that will vastly simplify many laborious recording and mixing functions and permit accurate recreation of specific EQ control set-ups.

- 4-band, dual-channel parametric equalizer. Each section offers overlapping tuning having a 25:1 frequency range; +16dB boost/−40dB cut in each band.
- Bandwidth variable from 5 to 0.1 octaves ("Q": 0.3 to 15).
- Proven Orban "Constant-Q" design enables use of equalizer as a true notch filter.
- Tunable 18dB/octave high-pass filter and 12dB/octave "Automatic Sliding Besselworth"™ low-pass filter provide maximum flexibility while preserving musicality.
- 99 non-volatile memory registers for instantaneous storage and accurate recall of complete control setups, including input gain.
- Digital displays show current settings of control parameters.
- High-quality, no-compromise audio path having no VCA's.
- Up to 14 two-channel slave units can be addressed by the master unit. Each pair of channels can be ganged to track in stereo or can be programmed independently.
- MIDI-controllable. Port for remote control. Provision for future serial interfaces.

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The New Orban 787A Programmable Mic Processor

Powerful, programmable processing for maximum impact.

Three-band parametric equalizer, compressor, de-esser, noise gate, and compressor gate integrated in a compact, powerful system. Stores up to 32 different control set-ups in memory for instant recall. Designed for mic processing and voice recording, but versatile enough for many other production uses. Optional MIDI interface.

- Line-level input with optional Jensen transformer mic preamp and 48-volt phantom power.
- 3-band parametric equalizer with variable frequency, bandwidth, and boost/cut for precision control.
- Smooth compressor delivers maximum presence and "punch" while maintaining consistent levels.
- Full-function de-esser helps control excessive sibilance.
- Noise gate attenuates noise by up to 25dB; compressor gate prevents rush-ups during pauses (front-panel control selects noise or compressor gate).
- Effects send-and-return with programmable return level simplifies integration of external reverb or "psychoacoustic exciter".
- STORE, RECALL, and COMPARE buttons provide instantaneous access to 32 user-programmed control setups.
- Digital display shows current settings of control parameters.
- Easy-to-read bargraph displays show output and gain reduction levels.
- Memory protected by internal back-up battery.
- Security code locks programming controls to prevent tampering.
- Built-in control connectors for remote control and MIDI or future serial interfaces.
- Provision for second-channel slave unit for dual-mono or stereo operation.

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The producer who moves from studio to studio knows that the final product can vary tremendously. The Programmable Mic Processor eliminates some of the uncertainty and variation, enabling you to get closer to your "sound" — regardless of where you are working! The professional artist can note the settings and the particular mic used for an especially satisfying session, and then use these as a reasonable starting point for processing when working in other studios or at live venues.

Installed Sound Systems and Live PA: Why use daisy-chained signal processing when one unit will do? The Programmable Mic Processor takes up less rack space, guarantees that headroom will be used optimally to minimize noise and distortion, and ensures consistent quality. Your talent can concern themselves with their performance, instead of worrying about the sound of the system.

Summary

Add Orban's new Model 787A Programmable Mic Processor to your arsenal. It delivers powerful mic and instrumental processing with an unprecedented combination of integrated, fully programmable functions. Instant access to 32 complete control setups enables you to achieve precise, consistent vocal and instrumental sounds day after day — while saving both time and money.

Estimated First Delivery: Fall '87

Estimated Price: Under $2000

PRELIMINARY (Subject to Change)

orban

The New Orban 222A Stereo Spatial Enhancer

An effective and affordable enhancement tool for on-air spatial definition.

- Proprietary, patent-pending technique detects and enhances psychoacoustic directional cues which are present in all stereo program material.
- Increases brightness, impact, and definition of music.
- Front-panel ENHANCEMENT and WIDTH LIMIT controls allow tailoring of processing to user requirements.
- No increase in FM multipath distortion, no unnatural exaggeration of reverberation, and no increase in sensitivity to vertical tracing distortion in disc playback.
- Full mono compatibility.
- Complements any broadcast audio processor without changing the station's "sound".
- Easy-to-read LED bargraph displays indicate status and degree of enhancement.

Why Stereo Spatial Enhancement?

In contemporary broadcast audio processing, high value is placed on the loudness and impact of a station compared to its competition. Our industry-leading OPTIMOD™ processors already have made a major contribution to competitiveness.

Now, the new Orban 222A Stereo Spatial Enhancer can augment your station's **spatial** image the way your audio processor maximizes loudness and brightness. Your stereo image will become magnified and intensified; your listeners will also hear more loudness, brightness, dynamics, and depth.

How You Use It & What It Does

The 222A has stereo inputs and outputs. It is designed to be inserted in the program line at the studio prior to processing. It dynamically changes the amplitude and phase content of the program material to increase apparent width, depth, and transient definition. It does not add "musical distortion". Intelligent gating makes the unit immune to small errors in channel balance, prevents over-enhancement, and avoids the "mushy", homogenized sound that has so often been the result with earlier techniques.

A Competitive Edge Without Nasty Side-Effects

There have been many attempts to process stereo signals to increase width and dimensionality by altering the ratio of the L−R signal to the L+R, or by adding controlled distortion (which sounds good on some program material but not on all.) The major problem has been that any enhancement of the L−R signal by prior techniques has always resulted in a significant increase in multipath distortion.

The 222A does not increase multipath distortion, exaggerate reverberation, or increase sensitivity to vertical tracing distortion in disc playback. It can be placed in the program line confidently with the assurance that it will provide the benefits of stereo spatial enhancement without the by-products which cause listener annoyance and tune-outs.

The Orban 222A Stereo Spatial Enhancer uses a new patent-pending technique which has been **extensively tested** in real-life, high multipath environments. These tests have resulted in uniformly rave reviews.

Most of All — Affordable Enhancement

In the refinement of your station's audio processing, you've no doubt become wary of equipment that's over-hyped, over-priced, and under-powered. The 222A Stereo Spatial Enchancer is **affordable**: It's price is only $995.00*. And it works.

Find out how well it works by trying it on the air, with your own format and your own processing chain. Contact your preferred Orban Dealer and arrange for a demo when the 222A becomes available. We think you'll find that our new enhancement tool is a powerful, competitive weapon that's just right for your image.

Estimated Price: $995.00*
(* Suggested List Price, USA)

First Delivery: January, 1988

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ORBAN
ACC-22
FM FILTER CARD
ACCESSORY FOR OPTIMOD-FM®

[] **Increases loudness: 6% increase in average modulation level.**

[] **Reduces SCA audio noise and data errors caused by main channel program-induced splatter.**

[] **Factory-installed option in new Model 8100A/1 OPTIMOD-FM.**

[] **Field Retrofit Kit for existing OPTIMOD-FM Models 8100A and 8100A/1.**

OPTIMOD-FM and SCA

OPTIMOD-FM Model 8100A uses overshoot-compensated lowpass filtering in its processing to assure that it meets all FCC Rules and Regulations regarding crosstalk. However, when aggressive processing and clipping are employed, there is a small amount of "splatter" into the SCA region caused by operation of the safety clippers. When the 8100A was designed, SCAs were used only for narrow dynamic range background music or voice, and the small amount of "splatter" was totally acceptable.

Today, many users of SCA are seeking far greater performance since SCAs are now transmitting wide dynamic range audio, data, and paging. In addition, processing has become far more aggressive in many markets, increasing the probability of "splatter."

The ACC-22 card reduces clipping "splatter" into the SCA region of the baseband by about 25dB, significantly reducing noise on audio material and error rates in data.

Loudness

In FM, loudness is limited by the requirement that peaks do not cause the carrier to exceed 100% modulation.

The ACC-22 reduces peak overshoots by about 6% (0.6dB), resulting in 6% additional modulation capability and a commensurate *increase in loudness* with no additional compression, limiting, or clipping--and no increase in processing artifacts.

Composite clipping has been used by some broadcasters in an attempt to clip overshoots and thereby increase loudness. Unfortunately, such clipping even in very small amounts dramatically increases trash in the SCA region. In addition, as clipping is increased to "bite" into program material below the overshoots, significant audible distortion is produced. For L+R material, the distortion increase is *identical* to that produced by simple audio clipping. To make matters worse, a live announcer (which is almost always pure L+R) is the program material most vulnerable to added clipping, even in small amounts.

Increases of fractions of a dB of composite clipping can cause baseband "trash" to increase by 10dB or more! To add insult to injury, composite clippers are expensive.

The additional modulation capability provided by the ACC-22 card can achieve loudness within *about 0.3dB* of that produced by composite clipping (an inaudible difference), *without any of the drawbacks*. In fact, with ACC-22 installed, overshoots are virtually eliminated--additional composite clipping can do nothing but add audible distortion and SCA trash.

Technical Description

The left and right audio paths in Model 8100A include overshoot-corrected lowpass filters followed by safety clippers. The ACC-22 replaces each safety clipper with two cascaded fifth-order overshoot-compensated lowpass filters.

These new filters improve the stock 8100A in several ways:

Overshoots are reduced by about 6% (0.6dB).

Program-induced trash above 61kHz is reduced to below -75dB referenced to 100% modulation, an improvement of about 25dB. (Where composite clipping had been used, program-induced trash can be reduced by up to 40dB.)

Dynamic main-to-sub and sub-to-main crosstalk performance is improved to the static limits of the stereo generator, as is dynamic separation.

Recommended Applications

All stations using SCA.

All stations who wish to eliminate their composite clipper without loudness penalty.

All stations using 8100A without a composite clipper who wish to improve loudness without suffering the side effects of composite clipping.

If your station does not broadcast SCA, and if you are not using composite clipping and are happy with the loudness levels achieved by the Model 8100A, the ACC-22 is not recommended.

A word of warning: Multipath in composite STLs introduces its own trash into the SCA region which could mask benefits expected of the ACC-22.

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Revision 09, Effective 1 April 1987
Supersedes Revision 08, 15 August 1986
Changes: Delete RET-004

ORDERING GUIDE & SUGGESTED LIST PRICES

OPTIMOD-FM BROADCAST PRODUCTS

Model **Description**

OPTIMOD-FM

Model	Description	USA List Price
	OPTIMOD-FM	
8100A/1	**OPTIMOD-FM AUDIO PROCESSING SYSTEM with STEREO GENERATOR** Complete on-air processing for FM broadcast. Includes dual-band stereo compressor, high-frequency limiter, smart clippers, built-in stereo generator that supplies a composite baseband output. 115/230V, 50-60Hz. 75us pre-emphasis is standard; order OPT-011 for 50us installed (no charge).	**$4,995.00**
ACC-022	**FM Filter Card option** Factory-installed option for 8100A/1. Supplied as a Retrofit Kit for existing units. Offers enhanced SCA protection and increased average modulation capability (by about 6%). Provides 25dB more protection to 67kHz SCA than provided by standard OPTIMOD-FM. Compatible with XT2 Six-Band Limiter.	**$595.00**
	FM ACCESSORY CHASSIS	
8100A/ST	**Studio Chassis** Separates OPTIMOD-FM into two chassis to locate compressors at studio. Controls average levels into STL or telephone/post lines, and optimizes signal-to-noise ratio. Works with 8100A or 8100A/1. Compatible with XT2 Six-Band Limiter. 115/230V, 50-60Hz.	**$895.00**
8100A/XT2	**Six-Band Limiter** Accessory to 8100A and 8100A/1. Improves loudness, brightness, and consistency compared to OPTIMOD-FM processing alone. Controls for BASS, PRESENCE, DENSITY, and CLIPPING let you accurately fine-tune the processing for your target audience. The earlier Model 8100A requires RET-027 Retrofit Kit; see below.	**$2,075.00**

See reverse side for FM ACCESSORIES

Prices are FOB ORBAN ASSOCIATES INC., SAN FRANCISCO, and are subject to change without notice. Orban Broadcast Products are sold through authorized Orban Broadcast Dealers worldwide. For names of Dealers near you, or for more information, call or Telex Orban.

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Model	Description	USA List Price
	FM ACCESSORIES	
RET-027	8100A Retrofit Kit to accept XT2 Six-Band Limiter Used to upgrade 8100A to 8100A/1, to accept 8100A/XT2 Six-Band Limiter. Includes replacement circuit card, prewired connector assembly.	$395.00
RET-039	Retrofit Kit to convert 8100A/XT to XT2 Six-Band Limiter	$49.00
ATE-3F	Interface Panel for Harris TE-1 or TE-3 Exciter	$75.00
RCA-1	Shorting Connector for RCA BTE-15 Exciter	$10.00

Note: For Continental 510R-1, Collins 310Z-1 and 310Z-2 exciters, obtain interface from Continental. Most other direct-FM exciters with broadband inputs do not require special interface.

PRELIMINARY BROCHURE

orban

The Orban 8100A/XT2 Six-Band Limiter

Accessory Chassis for OPTIMOD-FM®

- Increases loudness without audible side-effects.
- Improves consistency from source to source.
- Increases presence and intelligibility on auto and table radios.
- Provides control over bass, presence, brilliance, density, and clipping.
- Combines with your OPTIMOD-FM into a system that provides the best performance *and* value.
- Works with any 8100A-series OPTIMOD-FM.

Audio Processing for Position

The 8100A/XT2 OPTIMOD-FM System provides added *power* to fine-tune your air sound to your target audience and market position, through precise control of bass and treble sound texture, program density, and program dynamics.

It's for those stations that desire a *louder* or more processed sound than that provided by a "stock" 8100A OPTIMOD-FM set up according to our recommendations – a *louder* sound with *fewer* processing artifacts.

And it's an *all-Orban, all-OPTIMOD* system, so that all parts work together harmoniously.

It's the powerful alternative to processors that go in front of an OPTIMOD-FM. The XT2 goes in the *middle* of the OPTIMOD-FM, to give you *loudness*, *brightness*, and *clarity*, without the inevitable side-effects that result when other outboard processors are added.

The XT2 Sound

Suppose you have a "barefoot" OPTIMOD-FM on the air, set to its recommended settings. You're listening in your control room, comparing board "program" to your "air" sound. You'll notice that "air" is louder, the level more consistent. But the basic sonic *texture* is similar to the original program material. That's what OPTIMOD-FM is famous for.

Now you add the XT2. The sound is suddenly *louder* and *brighter*, yet still remarkably open. The bass has punch, impact. The mids are clean, clear, and smooth, with a dramatic sense of presence. The highs are in perfect balance to the mids and lows, always present, but never brittle.

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Your **music sound** has a remarkable consistency from cut to cut, from a golden oldie to the latest CD release. Never boomy or bass-shy, never shrill or dull.

Your **announcer's voice** has increased presence and intelligibility. Without even a *hint* of processing distortion. Your **news actualities** from low-grade telephone calls are much more intelligible.

That's the sound of the XT2 OPTIMOD-FM System – clean, crisp, and consistent.

Now compare your sound to your over-processed competitors. You're as loud or louder. You're probably cleaner. But most importantly, your sound is comfortably listenable for long periods of time. With the XT2, there's no listener fatigue to drive your listeners away.

A Competitive Weapon

When to mobilize it: The XT2 is at home with those formats that demand that "big-time, show-business" gloss in their sound: mass-appeal highly competitive pop formats, such as CHR, AOR, AC, Country, and Black/Urban. But the XT2 may not be appropriate for a format that demands a sound that's strictly faithful to the original recording – the "barefoot" 8100A does this beautifully.

Even so, the XT2 (when used lightly) works surprisingly well for some Beautiful Music formats. Such stations are often used as background music, and music played softly tends to lose its highs and lows. The XT2 provides the required boosts and maintains a consistent frequency balance from cut to cut.

And the XT2's consistency helps to make the frequency balance of records from different eras more consistent. Used lightly, it can usually augment any live disk operation.

Talk formats can similarly benefit from the XT2's consistency. It's invaluable in pulling low-grade phone calls and actualities out of the mud, in real time, without operator intervention.

For the Engineer

The XT2 consists of a six-band limiter cascaded with Orban's patented multi-band distortion-cancelled clipper. Functionally, the XT2 replaces the high-frequency limiter within the 8100A. When the XT2 is in use, the OPTIMOD-FM's dual-band compressor is converted into a slow averaging AGC in front of the six-band limiter to always operate the XT2 in its "sweet spot."

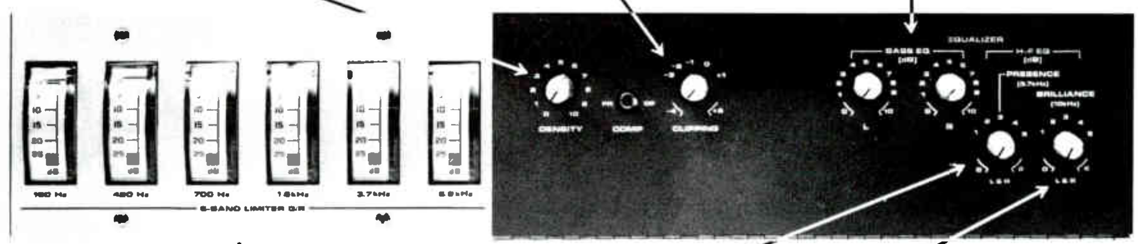
The result? Operator gain riding becomes almost completely non-critical, and the six-band limiter acts as an intelligent "automatic equalizer" that gives your station a consistent, easy-to-listen-to sound, source-to-source and cut-to-cut.

Why six bands? The 8100A already has a dual-band compressor that works fine as long as it is not stressed to produce large increases in program density. But when you *do* try to get such density with the 8100A alone, the sound gets fatiguing – certain program material produces audible and disturbing "spectral gain intermodulation." That is, instruments in one frequency range can cause audible changes in the loudness of instruments in another frequency range; the dominant instrument literally modulates the others.

DENSITY determines the input drive level to the Six-Band Limiter. Lets you have it your way—open and transparent, or solid and dense.

CLIPPING adjusts the drive level into the multiband clippers, determining the loudness/distortion tradeoff.

BASS EQ provides peaking boost at 65Hz, making it easy to get the solid punch you need for many contemporary music formats.



Gain reduction meters for each band: they provide the information you need for accurate setup.

PRESENCE boosts the 3.7kHz band to achieve midrange balances right for your format.

BRILLIANCE boosts the 10kHz band. Use it to increase the sense of "air" and "transparency" in your music.

THE NEW ORBAN 6-BAND FM LIMITER. (WE LISTENED.)

Many FM stations perpetually seek "the perfect sound". OPTIMOD-FM alone does it for many. The OPTIMOD XT Accessory Chassis improved results for some. Still, some seek even more from OPTIMOD-FM.

We listened.

Our *NEW* **8100A/XT2 Six-Band Limiter Accessory Chassis** (which works with any 8100A OPTIMOD-FM) features two new high-frequency equalizer controls: PRESENCE and BRILLIANCE. They complement the original 8100A/XT's bass EQ controls, and give you *twice the flexibility* of the single HF EQ control typical of other add-on multiband processors.

With an XT2, your OPTIMOD-FM system is totally immune to operator gain-riding errors because the dual-band compressor in the main unit is converted into a smooth, slow AGC to ride gain ahead of the XT2. Any reasonable input level operates the XT2 in its "sweet spot," so there's never any need to add external, potentially incompatible compression.

This is good news because the time-constants and other processing parameters in a pure, integrated Orban system have been carefully harmonized to achieve an overall sound that's *loud* and *bright*, yet remarkably *open* and free from audible side-effects.

The XT2 also excels in the most difficult of processing tradeoffs—delivering loudness on music while keeping speech free from clipping distortion. Credit this uniquely capable performance to Orban's patented multiband distortion-cancelled clipping system—which we were able to implement in the XT2 system because the XT's circuitry is fully *integrated* into the processing system, not just tacked onto the front.

The XT2 lets you have it all: natural sound, source-to-source consistency, loudness, clean voice, and adjustability that lets you tailor bass and treble to your taste and format requirements. And thanks to its efficient single-chassis construction and its use of the main 8100A power supply, it lets you have the next step in Optimod processing at an exceptionally reasonable price: $2075 (suggested list).

We listen to our customers. Listen to our new XT2. We think you'll like what you hear.

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To completely eliminate this problem, the 8100A/XT2 OPTIMOD-FM System achieves most of its density increase in the XT2's six-band limiter. Because each band in the XT2 processes only a small fraction of the total frequency range, the XT2 is completely free from spectral gain intermodulation. And the significant peak clipping is performed multi-band as well. The outputs of the individual bands are applied to their own clippers, which are followed by filters to reduce out-of-band harmonic and IM distortion.

This scheme is far more effective than a system consisting of an 8100A preceded by a compressor, wide-band or multi-band.

The XT2 cannot make a station any louder than an 8100A OPTIMOD-FM operated aggressively and driven with very bright program material. (No other add-on can do that either!) But the XT2's multi-band processing allows you to achieve those very high levels of loudness *without the processing artifacts* that add listening fatigue.

Getting the sound you want: The XT2 sounds great on music *and* voice. This is an important point when comparing the XT2 to other add-ons. Any add-on that goes in front of an OPTIMOD-FM relies on the OPTIMOD-FM's circuitry to avoid clipping distortion and to provide most of the density increase. And wide- or dual-band compression after multi-band compression cannot help but sound strained and pumpy on some program material (spectral gain intermodulation again!).

When setting up the OPTIMOD-FM for maximum loudness, a station is usually limited by how much live-voice distortion is tolerable – music will tolerate more clipping than voice. The 8100A/XT2 OPTIMOD-FM System controls live voice levels better than an OPTIMOD-FM alone. Therefore, it is usually possible to set up the processing to get the same high loudness on music as a very aggressively operated OPTIMOD-FM – without the increase in voice distortion that almost always accompanies such aggressive operation. Any add-on which goes in front of the OPTIMOD-FM cannot control voice distortion nearly as well.

The XT2 sounds best when its multi-band limiters are driven lightly – the sound is loud, bright, clean, easy-to-listen-to. When driven heavily, the sound gets more dense, but surprisingly, it doesn't get much louder. (Some like it that way, so the XT2 provides both options.)

Installation

The XT2 can be used with any 8100A-series OPTIMOD-FM, in both single and dual chassis configurations. It always resides at the main OPTIMOD-FM chassis, and talks to its host through a multi-pin connector and short interconnecting cable. It requires two rack units.

The Model 8100A/1 is ready to accept the XT2 without modification.

The earlier Model 8100A requires field-conversion to accept the XT2. The conversion consists of mounting a pre-wired multi-pin connector in an existing port in the rear panel, and soldering the wires to appropriate locations on the motherboard. Card #5 is replaced with an updated version that allows the multi-band compressor to be used as a slow AGC. Very minor modifications must be made to other cards. All parts and easy-to-follow conversion instructions are supplied in Orban's Retrofit Kit RET-27.

Precise Control to Fine-Tune Your Sound

The XT2 includes all the sound-tailoring adjustments that most station engineers desire to get the sound that's just right for your format. And the EQ controls do not affect the midrange sound that is so critical (and so hard to get right with most of the other multi-band units).

Density Increase in the 8100A/XT2 OPTIMOD-FM System is achieved by dual-band compression followed by six-band limiting. Moderate and subtle 8100A-style density increase coupled with light multi-band limiting achieves an open sound without compromising loudness or consistency.

RELEASE TIME	Varies the AGC from a slow "hand on the pot" to a density-augmenting dual-band compressor. Either way, the multi-band limiters in the XT2 are always driven in their "sweet spot."
DENSITY	Controls the amount of multi-band limiting, and lets you vary the texture from open and transparent to solid and dense – or split the difference. Even the "solid and dense" sound is free from pumping and spectral gain-intermodulation.
CLIPPING	Affects the loudness/distortion trade-off by controlling how hard the clippers are driven. With Orban's patented multi-band distortion-cancelling clipper, very effective reduction in peaks is achieveable without the distortion build-up typical of other clippers.

Bass Equalizers gives you separate control of "warmth" and "punch." Because the XT2 often increases the brightness of program material, some bass boost is usually desirable to keep the sound spectrally well balanced.

BASS COUPLING	Controls the amount of "warmth" added to bass-shy material, using the dual-band compressor as a dynamic low-frequency shelving equalizer below 200Hz.
BASS EQ	Provides a peaking boost (at 65Hz) to achieve solid, punchy bass from most decent consumer radios – while avoiding exciting the mid-bass dashboard resonances in cars.

High-Frequency Equalizers are carefully integrated into the design of the top two bands of limiting. This avoids the problem of "fixed" equalizers that sound just right on some program material and horribly wrong on others. Together, the PRESENCE and BRILLIANCE controls let you achieve the precise high frequency balance you want – no single treble equalizer can sound this good.

PRESENCE	Boosts the 2-6kHz region to achieve presence and loudness increases as desired. Yet its location "within the limiter" prevents excessive stridency with program material already having a great deal of presence energy.
BRILLIANCE	Boosts the region centered at 10kHz to provide an audibly attractive effect, increasing the "air" and "transparency" (similar to a psychoacoustic exciter). The six-band limiter is configured so that it doesn't "fight" BRILLIANCE boosts – the XT2 can achieve exceptional brightness if you want it.

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The basic clipper is followed by a 15kHz lowpass filter to assure freedom from aliasing distortion. In addition, difference-frequency distortion below 2.2kHz introduced by clipping is *cancelled* by our patented "Smart Clipper" frequency-dependent distortion-nulling technique. This permits much harder clipping without audible distortion (particularly at high frequencies) than was heretofore possible. The result is dramatically improved high-frequency power-handling capacity.

Frequency Contoured Sidechain (FCS) Overshoot Compensator: While all filter overshoots are minimized by the use of phase correctors (even in the preemphasis network!), the output of the clipper section (including 15kHz lowpass filter and distortion cancellation) contains substantial overshoots due to addition of the distortion-cancellation signal, and to unavoidable overshoots in the lowpass filter. These overshoots must be eliminated without introducing frequency components above 19kHz which would otherwise cause "aliasing" and nonlinear crosstalk in the stereo baseband. The overshoot compensation in the 8000A slightly limits the high frequency power handling capacity of the system. The overshoot compensation system of our major competitor suffers from an inordinate sensitivity to overdrive; it causes *much* more IM distortion than a simple clipper when both are overdriven equally.

We have developed a new "Frequency-Contoured Sidechain" (FCS) overshoot compensator for the 8100A which offers the "best of all possible worlds". It permits high modulation at all frequencies, yet does not suffer from excessive IM (compared to a simple clipper) when overdriven. Simultaneously, it offers extremely good suppression of out-of-band frequency components.

About The Stereo Generator

The stereo generator in the 8100A uses the "matrix" approach to generate the stereo composite signal, as opposed to the more common "switching" approach. It features feedback stabilization of pilot phase, pilot amplitude, *and* separation, optically isolated remote control of stereo/mono function switching, *and* extensive metering.

This high-performance generator features better than 60dB separation (typical), distortion below 0.02%, and superior audio quality because the L+R (dominant in most program material) is not subject to any switching or modulation process, unlike the latest "digital" stereo generators.

Because Part 73.322 of the *FCC Rules* refers to performance requirements for the *stereo generator, exciter, and RF amplifiers*, it is permissible to measure main-to-sub and sub-to-main crosstalk at the input terminals to the stereo generator. The 8100A stereo generator provides internal main-to-sub and sub-to-main crosstalk test modes: a switch permits injecting the output of the right channel processing into either the "main" or "sub" inputs of the stereo generator. An external test box is not required, making verification of cross-talk performance significantly more convenient.

Configuration and Installation

The availability of 25dB of gain reduction range means that the 8100A never needs an external AGC. At stations which install the 8100A at the transmitter and pass audio to the transmitter through an STL with limited signal-to-noise ratio, substantial benefits are achieved by applying compression prior to the STL. For this reason, the compression section of the 8100A can be installed in an Accessory Chassis to permit compression at the studio end.

The Studio Accessory Chassis is low in cost and houses only the dual-band compressor. Its use protects the STL from overload and provides control over most of the processing at the studio. Adjusting the gain between the two chassis is aided by calibrated metering, and requires only an audio oscillator for alignment.

If available, a *composite* STL will allow installation of the entire OPTIMOD-FM at the studio, with the 8100A's baseband connected directly to the STL transmitter.

In either configuration, practical requirements have been carefully considered. Operation is possible with input levels as low as −30dBm, and rigorous RFI shielding is employed. The stereo generator output is held floating over ground to avoid introducing system ground loops when unbalanced exciter inputs are used. An optically-isolated momentary remote switching facility for *stereo*, *mono-left*, and *mono-right* modes yields maximum versatility and freedom from RFI or ground-loop interference.

To facilitate proofs, a pair of PROOF/OPERATE switches defeat all compression and limiting, yet do not bypass any electronics normally employed in *Operate* mode. To facilitate crosstalk measurements, a NORMAL/MAIN-TO-SUB/SUB-TO-MAIN switch is provided on the stereo generator to establish the necessary test conditions without need for an external test fixture.

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SPECIFICATIONS

The following specifications are presented to satisfy the engineer that they seem reasonable, to help him plan his installation, to help him make certain comparisons with other familiar processing equipment with which he is familiar, to verify that the 8100A can readily pass a Proof of Performance, and to verify that it meets all requirements in the *FCC Rules*.

Years of experience listening to and designing audio processors have convinced us that there are no conventional specifications which correlate well to the listening qualities of a processing system like OPTIMOD-FM. Although it may sound strange to an engineer, we can state with confidence that, given the current state-of-the art in audio measurement techniques, the only way to meaningfully evaluate the distortion introduced by an audio processor is by *subjective listening tests.*

For this reason, no system distortion specifications are presented for the 8100A in *Operate* mode. In this mode, harmonic distortion is highly frequency-dependent, and correlates very well to a listener's actual impression of whether the harmonic distortion test tones emerging from the processor *sound* "distorted" if *listened* to on speakers or phones.

The CCIF Difference-Frequency IM test measures the amplitude of a low-frequency tone produced by the device under test when excited by two high-frequency tones. Because the distortion tone and test tones are far apart in frequency, the distortion tone would not tend to be well-masked by the test tones. Thus CCIF IM is a sensitive indicator of audible distortion—much more so than the more commonly-used SMPTE IM test.

Compared to a simple clipper, the patented distortion-cancelling circuitry in the 8100A "Smart Clipper" is most effective in reducing CCIF IM—thus improving listening quality in a way that does not correlate to more commonly used measurements.

FREQUENCY RESPONSE
(System in *PROOF* mode)
Follows standard 75us preemphasis curve ±0.75dB, 50-15,000 Hz. 50us preemphasis available on special order. All preemphasis networks include a fourth-order lowpass filter and fourth-order phase corrector prior to the high-frequency limiter and clipper to prevent these elements from processing out-of-band program material and to minimize overshoot, thus minimizing the amount of high-frequency limiting and clipping. (NOTE: The Dolby 334 Broadcast Encoder, which is compatible with the 8100A, internally transforms 75us to 25us. Thus broadcasters wishing to use Dolby-B encoding should order standard 75us preemphasis along with the optional Dolby connector.)

INPUT CONDITIONING
Highpass Filter: Third-order Chebychev with 30Hz cutoff and 0.5dB passband ripple. Down 0.5dB at 30Hz; 10.5dB at 20Hz; 31.5dB at 10Hz. Protects against infrasonic destabilization of certain exciters' AFC's, as well as infrasonic gain modulation in the compressor.
Phase Scrambler: Allpass network makes peaks more symmetrical to best utilize the symmetrical peak overload characteristics of the FM medium.

NOISE
−75dB below 100% modulation, 50-15,000 Hz maximum; −81dB typical.

TOTAL SYSTEM DISTORTION
(*PROOF* Mode; 100% Modulation)
Less than 0.05% THD, 50-15,000Hz (0.02% typical); less than 0.05% SMPTE Intermodulation Distortion (60/7000Hz; 4:1).

"MASTER" BAND COMPRESSOR CHARACTERISTICS
Attack Time: approximately 1ms
Release Time: program-controlled—varies according to program dynamics and amount of gain reduction (see text). Process can be scaled fast or slow by means of continuously variable RELEASE TIME control. Employs delayed release for distortion reduction.
Total Harmonic Distortion (measured at VCA output, OPERATE Mode, RELEASE TIME control centered): Less than 0.1%, 50-15,000Hz, 0-25dB gain reduction
Available Gain Reduction: 25dB
Metering: Three dB-linear edgewise-reading gain reduction meters—
MASTER is true peak-reading with electronic acceleration and peak-hold (0-25dB);
COMPRESSOR indicates slow compression component of gain reduction (0-25dB)
LIMITER indicates fast peak limiting component of gain reduction (0-5dB)
Gain Control Element: True VCA. Proprietary Class-A design eliminates crossover notch distortion, modulation noise, and slewrate limiting found in competitive Class-AB designs.

"BASS" BAND COMPRESSOR CHARACTERISTICS
Attack Time: program-controlled; not adjustable.
Release Time: program-controlled; not adjustable. Incorporates delayed-release distortion reduction.
Total Harmonic Distortion (at VCA output, OPERATE mode): Less than 0.1% THD, 50-200Hz, 0-30dB gain reduction.
Available Gain Reduction: 30dB
Metering: single dB-linear edgewise-reading gain reduction meter (0-30dB).
Gain Reduction Element: Proprietary Class-A true VCA
Bass Coupling (U.S. patent #4,249,042): Enables gain of "Bass" band to track gain of "Master" band to any degree, from identical tracking to fully independent operation. Adjustable with BASS COUPLING control.

CROSSOVER CHARACTERISTICS
Control: 6dB/octave @ 200Hz;
Program: 12dB/octave @ 200Hz in unique "distributed crossover" configuration (U.S. patent #4,249,042)

HIGH FREQUENCY LIMITER CHARACTERISTICS
Attack Time: approximately 5ms
Release Time: approximately 20ms. Delayed release included for distortion reduction.
Mode: Left and right channels operate independently to avoid high frequencies in one channel causing audible timbre modulation of opposite channel.
Control Element: Junction FET
Metering: Two LED's indicate HF limiting in L and R channels.
Threshold of HF Limiting: User-adjustable over 3dB range to meet format requirements

FM "SMART CLIPPER" OUTPUT PROCESSOR CHARACTERISTICS
Nominal Bandwidth: 15.4kHz
Distortion Cancellation: Clipping distortion (below overshoot compensator threshold) cancelled better than 30dB (40dB typical), 20-2200 Hz (U.S. patent #4,208,548).
Delay Correction: Fourth-order allpass
Amount of Clipping: User-adjustable over 6dB range to match format requirements.

FREQUENCY-CONTOURED SIDECHAIN (FCS) OVERSHOOT COMPENSATOR CHARACTERISTICS (patent pending)
System Overshoot: The FCS circuit is best thought of as a "bandlimited safety clipper". It operates like a hard clipper, but does not produce out-of-band frequency components as a simple hard clipper would. Because the audio processing will sometimes limit steady-state material with high average energy (like sinewaves) or with very little high-frequency energy to levels below the threshold of clipping, it is difficult to state a clear and meaningful specification for the *system* overshoot performance of the FCS circuit. The FCS circuit is followed by a safety clipper. The overshoot specification could be slightly improved if this safety clipper were set up to clip more frequently. However, the system is aligned at the factory such that the safety clipper is almost *never* active, thus fully preserving the bandlimiting provided by the FCS circuit. With this safety clipper alignment, the peak modulation will be controlled ±3.5% on arbitrary waveforms clipped to any degree by the FCS circuit (acting as a bandlimited safety clipper); peak modulation will not exceed this level on other material. With typical program material, peak modulation uncertainty is less than 2%.
Sinewave Modulation Ability: 93% modulation (i.e., 0.6dB below maximum overshoot level) at all sinewave frequencies, assuming sinewaves are applied to FCS input.
Dynamic Separation: better than 45dB
Difference-Frequency Intermodulation: FCS circuit causes no more audible IM (such as sibilance splatter) than would a simple hard clipper clipping to the same depth. The entire 8100A processing system is specifically configured to prevent the FCS circuit from audibly degrading the difference-frequency distortion-cancellation properties of the earlier FM "Smart Clipper".

SYSTEM SEPARATION
Greater than 45dB, 50-15,000Hz; 60dB typical

STEREO GENERATOR CHARACTERISTICS
Crosstalk (Main Channel-to-Subchannel, or Subchannel-to-Main Channel): better than −40dB, 50-15,000Hz as measured at input terminals to stereo generator, or using internal crosstalk test mode which applies right-channel
generator inputs. Crosstalk representing distortion components (non-linear crosstalk) typically better than −80dB as measured on a baseband spectrum analyzer.
38kHz Subcarrier Suppression: Greater than 40dB below 100% modulation; 60dB typical
Suppression of 76kHz and its Sidebands: Greater than 70dB below 100% modulation
Pilot Frequency: 19.000kHz ±2Hz
Pilot Injection Adjustment Range: Less than 8% to greater than 10% modulation

INPUT
Impedance: greater than 10K ohms, electronically balanced by means of true instrumentation amplifier. Requires balanced source.
Common Mode Rejection: Greater than 60dB @ 60Hz
Sensitivity: −10dBm produces 10dB "Master" Band gain reduction @ 1kHz. Removal of internal 20dB pad permits −30dBm to produce same effect.
Connector: Cinch-Jones 140-style barrier strip (#5 screw).

COMPOSITE (BASEBAND) OUTPUT
Source Impedance: 470 ohms, independent of OUTPUT ATTEN setting, unbalanced.
Level: variable 0 to greater than 4V p-p by means of 15-turn OUTPUT ATTEN control
Connector: Type BNC held floating over chassis ground to permit interface to various exciters without need for wideband transformer for ground loop suppression. RF suppressed.
Recommended Maximum Cable Length: 6 ft (1.8m) RG-58A/U

AUXILIARY INPUT/OUTPUT (for Test use only)
Provides L and R lowpass filter output or L and R stereo generator input depending upon setting of rear-apron NORMAL/TEST switch. Connectors are RCA phono-type, unbalanced. Stereo generator requires approx. 3V RMS for 100% modulation, unbalanced, with source impedance of test generator less than 50 ohms.

OPERATING CONTROLS
VU Meter Selector: switches VU meter to read:
L or R Input Buffer
L or R Compressor Out
L or R Filter Out
L-R Level
19kHz Oscillator Level
38kHz PLL Control Voltage
38kHz AGC Control Voltage
±15 V Power Supply Voltages
Stereo/Mono Mode Switch: Momentary front panel switch may be conveniently strapped for either *left* or *right* mono by means of a plug-in internal jumper. Mode may be remote-controlled by application of 6-24 V AC or DC pulses to appropriate rear terminals. Terminals are optically isolated, and may be floated ±50 V above ground. Three pairs of remote terminals will select either *left* or *right* audio inputs in *mono* mode, or *stereo*. Another internal jumper selects which of the three modes will be entered on powerup.

SETUP CONTROLS (front-panel, behind lockable swing-down door)
Compressor:
Left and Right Input Attenuators
"Master" Band Release Time
Gate Threshold
Bass Coupling
Clipping
High-Frequency Limiter Threshold
Stereo Generator:
Pilot Injection
Pilot Phase
L-R Gain (Separation)
Pilot ON/OFF Switch
NORMAL/MAIN-TO-SUB/SUB-TO-MAIN Crosstalk Test Switch (see text)
General:
Output Attenuator
PROOF/OPERATE Switches (to defeat gain reduction, HF limiting, clipping, and gating)
Power ON/OFF
115V/230V Selector Switch

POWER REQUIREMENT
115/230VAC, ±15%, 50-60Hz, approx. 19VA. IEC mains connector with detachable 3-wire "U-Ground" power cord supplied. Leakage to chassis less than 0.25mA @ 115VAC; 0.5mA @ 230VAC.
AC is RF-suppressed.

DIMENSIONS
19" (48.3cm)W x 7" (17.8cm)H x 12.5" (31.2cm)D—4 rack units

SHIPPING WEIGHT
35 lbs; 15.9 kg

ENVIRONMENTAL
Operating Temperature Range: 0-50 degrees C (32-122 degrees F).
Humidity: 0-95% R.H., non-condensing

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The Multiband OPTIMOD-FM®

Performance Highlights

- ☐ Multiband compressor/limiter/ stereo generator
- ☐ Multiband *or* wideband operation plus versatile setup controls permit precise "tuning" for different formats
- ☐ Freedom from processing artifacts and distortion
- ☐ Optimum voice/music balances
- ☐ Excellent high frequency power handling for brightest sound
- ☐ State-of-the-art stereo generator
- ☐ Overshoot compensator permits full modulation at all frequencies, yet doesn't increase low frequency IM distortion

Plus

- ☐ Dual-chassis (studio/transmitter) option
- ☐ Excellent stability and unit-to-unit uniformity
- ☐ Plug-in card construction for easy maintenance
- ☐ Built-in crosstalk test generator
- ☐ Quad coupling ability
- ☐ Rigorous RFI shielding
- ☐ True VCA gain control
- ☐ True peak-reading gain reduction meter
- ☐ Low stereo generator output impedance permits longer baseband cable runs
- ☐ Orban-quality construction, documentation, and backup support

What It Will Do For Your Sound

The Orban OPTIMOD-FM Model 8100A is ideal for any format, and is the best-sounding FM processor that Orban knows how to make. As will be explained in more detail below, it can be operated in either "wideband" or "multiband" modes. Its advantages will be particularly appreciated in formats that ordinarily use heavy processing and that play recently-recorded program material with large amounts of high- and low-frequency energy. Such stations will find that 8100A "heavy processing" is free from the pumping, gain modulation, distortion, and fatigue that many have associated with such processing in the past.

Stations that prefer lighter processing, but that play music with high transient content (such as rock, soul, disco/dance, or jazz), will find that the 8100A (adjusted for slow release times and "multiband" operation) permits use of much more compression than might be expected. Experience with other processors leads many to associate large amounts of compression with highly audible processing—the 8100A will shatter this preconception! And, whether processing is heavy or light, the super-sophisticated release-time circuitry requires much less accurate D.J. gain riding than virtually any other competitive processing.

Operated in "wideband" mode, the 8100A sounds similar to our previous 8000A in formats such as "beautiful music" that ordinarily use light processing and play relatively undemanding program material. The only difference is that processing is even smoother, and no high frequency loss is *ever* apparent on such material. Given the acceptance of the 8000A in such formats, this "family resemblance" assures that use of the 8100A will yield an even more non-fatiguing, audience-pleasing sound—a sound that yields high quarter-hour maintenance and which has resulted in outstanding ratings for so many 8000A "beautiful music" stations.

We are extremely gratified by the 8000A's acceptance among FM broadcasters, and have tried in this design to preserve the 8000A features which accounted for its popularity. We feel that these features include *simplicity of concept, relatively foolproof installation, loud, clean, non-fatiguing sound,* and a "feature" which has little to do with the product per se: a long-term company commitment to quality, reliability, and customer service.

At the same time, we have tried to respond to the demands of the current marketplace to produce a processor which can, when adjusted to do so, produce a "highly-processed" sound which is free from the usual compromises.

The result is a true second-generation overshoot-controlling FM processor. It is at once refined, sophisticated, easy to use, and extremely cost-effective. We believe that, given its versatility and superb sound, it is clearly the best processing investment you can make—and will enhance your quality and ratings now, and in years to come.

We look forward to your giving us the opportunity to show you what the new OPTIMOD-FM Model 8100A can do for *your* station or group.

The following sections describe in some detail the thinking and improvements that went into the new OPTIMOD-FM. We hope that you'll find this information interesting and useful in helping you make decisions about the kind of processing you'll need to stay competitive in the '80's. And we hope to show you how Orban can help you to achieve the ultimate competitive edge.

About the Wideband/ Multiband Compressor

Many variations of the traditional parallel-band triband processor have appeared. In our opinion, they all fail critical listening tests because the high frequency band can increase treble uncontrollably, resulting in a phony-sounding high end. Therefore, the 8100A employs a triband structure with a significant difference. The basic compressor consists of two bands—"Master" (above 200Hz), and "Bass" (below). After those two bands are summed together, the combined signal is preemphasized to follow the FM preemphasis curve, and *then* fed into a high-frequency limiter. Unlike the third band of a typical triband compressor, this limiter is very fast, and works only as hard as necessary to avoid audible distortion in the "FM Smart Clipper" following it (discussed below).

Because the "Smart Clipper" can clip far harder than the 8000A could without audible distortion, much less high-frequency limiting is required—and a new level of brightness and high-frequency power handling capability is achieved. The limitations of the 75us preemphasis curve are almost never apparent—even on very bright program material. In the 8100A, the brass retains its "bite"; the cymbals "sizzle". And the "phony highs" phenomenon is entirely absent.

Our dual-band compressor is unique in that it can be operated discriminately (with the bands independent of each other), or wideband.

The "independent" approach is most appropriate for "Top-40", "AOR", "Black", "Disco/Dance", and other formats emphasizing music containing heavy bass and dominant transient material. Program Directors programming these formats usually demand high loudness, high density, and considerable compression. The "independent" approach is ideally suited to these requirements because the requisite midrange density can be achieved without compromising bass definition and punch.

On the other hand, "Beautiful Music", "Fine Arts/Classical", and even some "MOR" stations cannot tolerate the frequency imbalance resulting from the "independent" approach. They often wish to use small amounts of compression and little or no density augmentation to preserve musical values and to avoid long-term listener fatigue. If the music or other program material typical of these formats contains passages without bass, then rumble and noise can be pumped up to an objectionable extent as the "Bass" band attempts to "fill in" with bass that doesn't exist except as noise.

To fill the needs of such formats, a patented BASS COUPLING control permits the "Bass" band to track the "Master" band at all times, except when extremely heavy bass appears at the input. Rather than causing audible gain modulation (as in a *true* wideband system), such bass momentarily causes the gain of the "Bass" band to fall below the gain of the "Master" band. Thus wideband-mode users have the best of both conventional wideband and "independent" approaches. With ordinary program material, the system operates wideband and frequency balances are preserved. If strong bass comes along, the system momentarily becomes "independent" to avoid wideband modulation effects.

The bass coupling is continuously variable between fully independent and fully wideband (retaining excess-bass control) to suit the needs of your market and tastes.

Regardless of the amount of bass coupling, exceptional smoothness is assured by "Smart" attack- and release-time characteristics in all bands. To complement the sophistication of the release-time characteristic, available gain reduction is 25dB in the "Master" band and 30dB in the "Bass" band—an external compressor is *never* required. To make this range usable without "breathing" or "noise rush-up" during pauses, the compressor is gated and "freezes" its gain when the input drops below an adjustable threshold.

The result is an amazingly natural quality—the sort of sound that can hold a listener quarter-hour after quarter-hour—combined with high loudness, superbly appropriate balances between voice and music, a singular absence of perceived distortion, and a new zenith in brightness and high-frequency power handling capability.

About The FM "Smart Clipper"

Loudness is principally dependent upon the design of the peak limiter, lowpass filter, and overshoot compensator. A delicate balance must be maintained between peak/average ratios attained, perceived distortion, and integrity of the baseband spectrum. The 8100A uses a newly designed peak limiting system in which peak limiting, bandwidth limitation, and overshoot control are all elegantly interrelated.

Unlike the questionable "composite clipper" approach which uses brute-force clipping of the stereo composite baseband signal (with attendant aliasing distortion and compromises in dynamic separation), the overshoot compensation schemes used in all OPTIMOD-FM's (including the original 8000A) maintain total integrity of the baseband spectrum. The processing never introduces nonlinear crosstalk, pilot modulation, or other problems inherent in simpler approaches.

Peak Limiting and Distortion Cancellation: To preserve the naturalness achieved in the multiband compressor, peak limiting is performed by a clipper whose transfer curve has been carefully adjusted to yield minimum perceived distortion in conjunction with the balance of the entire 8100A system. Special circuitry throughout the multiband compressor section assures that the output of the compressor can be applied directly to the clipper—no broadband gain reduction is needed for distortion control. Voice is clean—yet music is loud. And voice/music balances are ideal.

cleaner, brighter, louder

FREQUENCY RESPONSE (System in TEST mode)
Follows standard 75 μS pre-emphasis curve ± 1 dB, 50-15,000 Hz. 50 μS and 25 μS available on special order.

HIGH PASS FILTER
3rd order Chebychev with 30 Hz cutoff. Down 0.5 dB at 30 Hz; 10.5 dB at 20 Hz; 31.5 dB at 10 Hz. Protects against subsonic destabilization of some exciters' AFC's. Can be strapped out.

NOISE
–75 dB max; –80 dB typical (50-15,000 Hz through 75 μS deemphasis).

DISTORTION, TOTAL SYSTEM
0.3% THD max, 50-15,000 Hz with any degree of gain reduction; 0.1% THD typical. In TEST mode (Instantaneous limiters & AGC defeated), below 0.05% typical.

BROADBAND LIMITER CHARACTERISTICS
Attack Time: Approx. 2 ms for 10 dB gain reduction. Release Time: Program controlled by means of quadruple time-constant release time analog processor. Release time may be scaled fast or slow by means of continuously variable Release Time control available to user. Gain Reduction Range: At least 15 dB.

HIGH FREQUENCY LIMITER CHARACTERISTICS
Attack Time: Approx. 3 ms. Release Time: Varies around 15 ms according to program material.

SYSTEM SEPARATION
Better than 40 dB, 50-15,000 Hz. Typically 50 dB or better overall.

CROSSTALK
(Main Channel to Subchannel or Subchannel to Main Channel) Better than –40 dB, 50-15,000 Hz, as measured at input terminals of stereo generator per interpretation of Part 73.322 of FCC Rules. Crosstalk representing distortion components is typically better than –70 dB, as measured on a baseband spectrum analyzer.

38 kHz SUBCARRIER SUPPRESSION
–40 dB minimum, –55 dB typical.

SUPPRESSION OF ALL SPURIOUS EMISSIONS in 67 kHz SCA REGION
Better than –70 dB.

MODULATION CONTROL
System will overshoot no more than 3% with any program material whatever.

PILOT FREQUENCY
19 kHz ± 2 Hz, 0-50° C.

PILOT INJECTION ADJUSTMENT RANGE
Less than 8% to greater than 10%.

INPUT
Impedance: 600 ohms balanced and floating, RF suppressed. Level: –10 dBm produces 10 dB gain reduction with Input Attenuator controls full CW. Removal of internal 20 dB pad permits -30 dBm to produce the same effect. Connector: Cinch-Jones 140 style barrier strip (#5 screw).

COMPOSITE OUTPUT
Impedance: 0-1250 ohms dependent on setting of Output Level control; unbalanced. Level: 4 volts peak-to-peak max., continuously variable by means of 10-turn Output Level control available to user. Connector: Type BNC held floating over ground which permits interface to various exciters without use of wideband transformer and without creation of ground loops. RF suppressed. Cable: Max. length recommended: 24" (61 cm).

AUXILIARY INPUT / OUTPUT (for test use only)
Provides L and R lowpass filter output or L and R stereo generator input depending upon setting of rear apron NORMAL/TEST switch. Connectors are RCA Phono type, unbalanced. Stereo Generator requires approx. 3.0 V RMS for 100% modulation, unbalanced, with a source impedance of less than 50 ohms.

OPERATING CONTROLS
Meter Selector: Stereo/Mono Mode.
Mode may be remote controlled by application of 6 to 24 V AC or DC pulses to appropriate rear terminals. Terminals are optically isolated, and may be floated ± 50 volts above ground. Three pairs of remote terminals will select either left or right audio inputs in mono mode, or stereo. Front panel switch may be strapped for either L or R mono.

SETUP CONTROLS (Front Panel, behind security cover)
Left and Right Input Attenuators: Release Time: Output Attenuator: Pilot Level; Pilot Phase; L-R Gain; Test/Operate Switch; Pilot On-Off Switch. Test/Operate Switch defeats all compression and instantaneous limiting in TEST position.

INDICATORS
Power On is indicated by green LED driven by unregulated negative DC supply. Overload is indicated by red LED which lights if operator attempts to exceed maximum achievable gain reduction. Meter (VU scale and characteristics) reads L and R input levels, L-R level, broadband gain reduction, 19 kHz oscillator level; 38 kHz gain control voltage; 38 kHz phase control voltage, ±15 VDC regulated power supply busses. The gain reduction metering signal is available on the rear apron for remote application. +5 VDC corresponds to 0 dB gain reduction.

POWER REQUIREMENT
115/230 VAC, ±15%, 50-60 Hz, approx. 12 watts. 3 prong, U-ground power cord attached. AC is RF suppressed.

DIMENSIONS
19" (48.3 cm) wide x 3.5" (8.9 cm) high x 9.25" (23.5 cm) deep behind panel. Allow 2.5" (6.4 cm) additional depth for connections.

WEIGHT
Approx. 13 lb. (5.9 kg) Net; 20 lb. (9.1 kg) packed.

OPERATING TEMPERATURE RANGE
0-50° C.

WARRANTY
One year, parts and labor. Subject to limitations set forth upon our standard warranty agreement.

All specifications subject to change without notice.

The ORBAN/BROADCAST OPTIMOD 8000A is recognized by the FCC for broadcast use when interfaced to a direct-FM exciter according to instructions provided by Orban Associates Division. This may require purchase of a wideband interface from the exciter manufacturer.

ORBAN/BROADCAST brand products are marketed and sold exclusively through

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Marketing and Sales Agent
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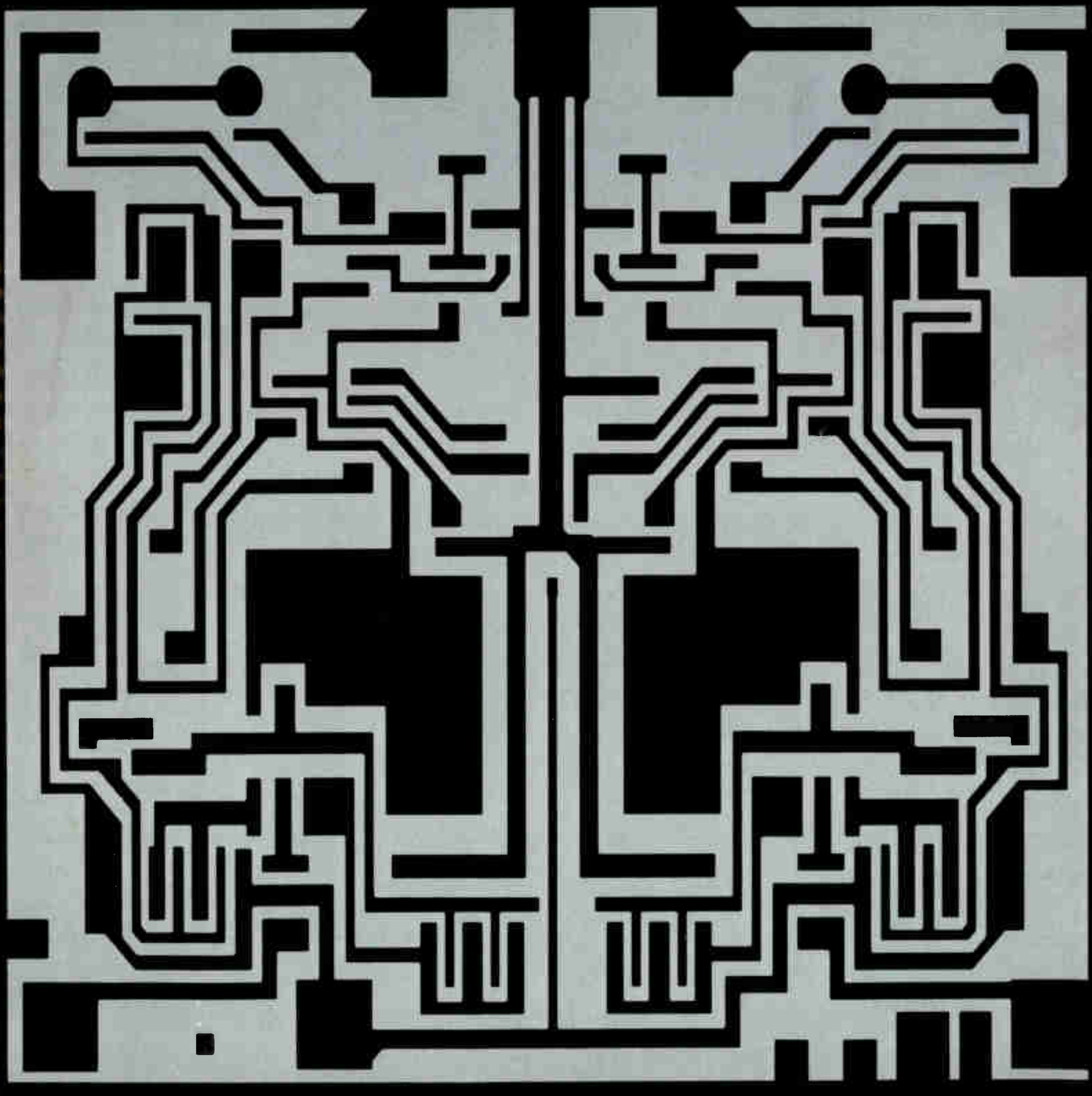
San Francisco, California 94133
(415) 421-3894

orban/broadcast

the systems approach to optimum FM modulation

OPTIMOD-FM



it would have made Major Armstrong very happy...

The most accurate fm limiter you have ever heard

OPTIMOD-FM

Ever since the FM market became competitive, FM broadcasters have been searching for ways to increase the quality and punch of their signals. Many audio processing devices have been introduced which purport to accomplish this goal. Without exception, the loudness gains obtainable with these devices bring a concurrent loss in audio quality which is easily perceived even on receivers of moderate quality. Gritty clipping distortion, a dull high end, and an uneasy pumping quality were the price to be paid for loudness.

After six years of research, ORBAN/BROADCAST has finally discovered a dramatically effective solution to the loudness/quality dilemma. This solution is now commercially available to broadcasters as the new Orban/Broadcast OPTIMOD-FM.

The OPTIMOD-FM is the result of a careful examination of every accepted principle of FM audio processing, and its effectiveness is due to a whole series of novel technical developments. The most revolutionary of these is OPTIMOD's system concept: the compressor, limiter, and stereo generator are engineered as a single system and incorporated in a single package.

BROADBAND LIMITING

Orban/Broadcast has thrown away conventional compressor/limiter design. Rather than diode attenuators with their high distortion, OPTIMOD utilizes clean FET attenuators. The FET's, combined with a highly developed delayed-response circuit, assures a total system harmonic distortion of typically less than 0.25% under any gain reduction condition.

Broadband gain control is accomplished with a single AGC amplifier which utilizes novel release-time computation circuit to perform both compression and limiting functions simultaneously. The relase time characteristics match the psychoacoustical properties of the ear far more closely than do older designs. Time constants and other release characteristics have been fine-tuned through hundreds of hours of critical listening under acoustically calibrated studio monitor conditions. The result is simply stated: The OPTIMOD-FM broadband AGC is the most subtle, most musical gain control ever offered to broadcasters. Eliminated are not only gross pumping, thumping, breathing, and other obvious misbehavior, but also the small gain shifts and musical miscalculations which cause the output of even the best of the older compressor/limiter combinations to sound audibly "processed". Yet OPTIMOD-FM is as loud or louder than these unmusical squashers and squeezers.

THE LIMITER CIRCUIT

Because FM broadcasters must employ 75 microsecond pre-emphasis to match conventional receivers, overmodulation due to excessive high frequency energy must be avoided. Traditionally, such control results in either distortion or audible loss of high frequencies. OPTIMOD employs a newly designed high frequency limiter which results in no audible high frequency loss with at least 90% of modern recordings, yet is so clean that it introduces no audible distortion to even studio master tapes. Because of the attack and release time circuitry, high frequency loss (in the cases where it occurs) sounds perfectly natural – without the rapid pumping in and out of high frequency information characteristic of older designs – and can be detected only by direct comparison with the original program material.

As a result of this rethinking of limiter/compressor design, the OPTIMOD-FM is essentially transparent – the output sounds exactly like the input, and varies only in volume – as if a super-skilled human operator was operating a fader. The relatively large amounts of peak limiting which occur are so complementary to the characteristics of the ear that this limiting simply isn't heard. Yet, complemented by the OPTIMOD lowpass filter, it accomplishes its goal of achieving loudness which is audibly superior to conventional systems with most program material.

CONTROL OF FAST-PEAK OVERMODULATION

Orban/Broadcast research has discovered that the principal cause of fast peak overmodulation – which has no effect on loudness, but which forces the broadcaster to lower his average level – is the lowpass filter incorporated in the input of every stereo generator of reputable manufacture. This lowpass filter is necessary to prevent interference between audio and the stereo pilot tone, and to avoid leakage between the main channel and stereo subchannel. Such filters are very sharp, and characteristically exhibit phase shift and ringing problems which introduce fast peaks as much as 30% to 40% above steady-state values – even if the limiter holds the peak levels at the input to the filter to precisely 100%. Because of the ringing and phase shift, the output level of the limiter must be lowered 2 to 3 dB in order to avoid illegal overmodulation on fast peaks.

Orban/Broadcast engineering has succeeded in designing a lowpass filter with the proper frequency response which overshoots a maximum of 3%, rather than the 30-40% of conventional filters. Therefore, peak modulation control in the OPTIMOD is "brick-wall", without the sloppiness of conventional systems, and average modulation levels can be raised 2 to 3 dB. This gain in loudness is accomplished totally by eliminating sloppiness, rather than by further increases in compression and limiting. Thus, audio quality is not degraded. In addition, the tight peak control of OPTIMOD means no more insecurity over where to set modulation to avoid an FCC citation.

STEREO GENERATOR

The OPTIMOD stereo utilizes a Gilbert-linearized multiplier to generate the L-R subchannel directly. This subchannel is then summed with the left and right audio signals and with the 19 kHz pilot to form the composite stereo signal. As opposed to the conventional switching approach, this highly refined matrix approach offers superior separation across the audio band, as no composite lowpass filter is necessary. High frequency intermodulation distortion is outstandingly low. To assure proper 67 kHz SCA operation, spurious outputs in the SCA region have been held below –70 dB. The circuitry is so clean, quiet, and precise that distortion, noise, and separation are limited entirely by the performance of the preceding audio processing chain. Stability with time and temperature are assured by controlling pilot level, pilot phase, and L-R/L+R gain with feedback loops.

To secure reliability and compact size, extensive use is made of IC technology. Extraordinary effort has been expended in suppressing potential RF interference, assuring trouble-free installation.

INSTALLATION

The composite output of the OPTIMOD has been designed to look like the output of a composite STL receiver. All major exciter manufacturers can supply interface devices for such receivers. In addition, Orban/Broadcast has successfully interfaced the OPTIMOD to a number of popular exciters directly. We will gladly discuss your particular requirements with you personally.

To facilitate installation directly into transmitter cabinets, a 230 volt connection is available on the power transformer.

Minimum required level for 10 dB gain reduction is –30 dBm, assuring satisfactory operation even if telephone line level is lower than normal.

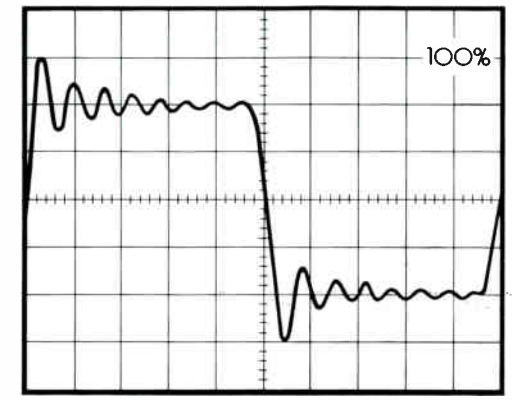
The OPTIMOD-FM has been designed to replace compressor, limiter and stereo generator. In the case of musical programming, further audio processing will only degrade quality!

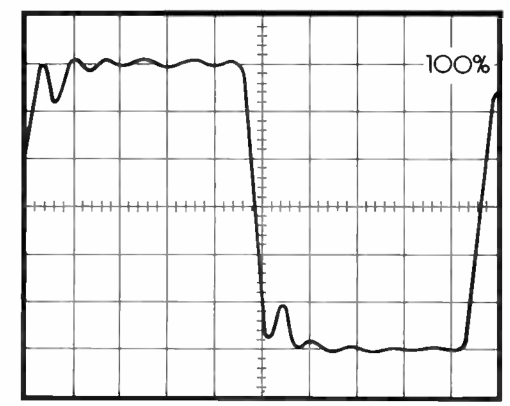
Setup is outstandingly easy. A normal level mono signal is sent from the studio, and the input level controls are adjusted for desired gain reduction and L-R null, both of which may read on the OPTIMOD's integral meter. The output attenuator, a ten turn control, is adjusted for the desired modulation. Pilot level is adjusted using the station's modulation monitor, while L-R gain and pilot phase are adjusted using oscilloscope patterns in the conventional manner. All these controls are accessible from the front panel, and are protected by a screw-on security cover.

OPERATION

Broadband gain reduction has purposely been limited to 15 dB. This assures that excessive compression cannot be used. In addition, noise expansion is limited to a maximum of approximately 10 dB, which compares favorably to older compressors which are ostensibly gated to eliminate noise build-up, yet can pull up noisy low-level passages 15-20 dB before gating occurs. On-air experience to date indicates that a 15 dB gain reduction range is entirely adequate for normal operation.

Stereo/mono switching is accomplished with a front-panel momentary switch, and opto-isolated remote control of this function is available.





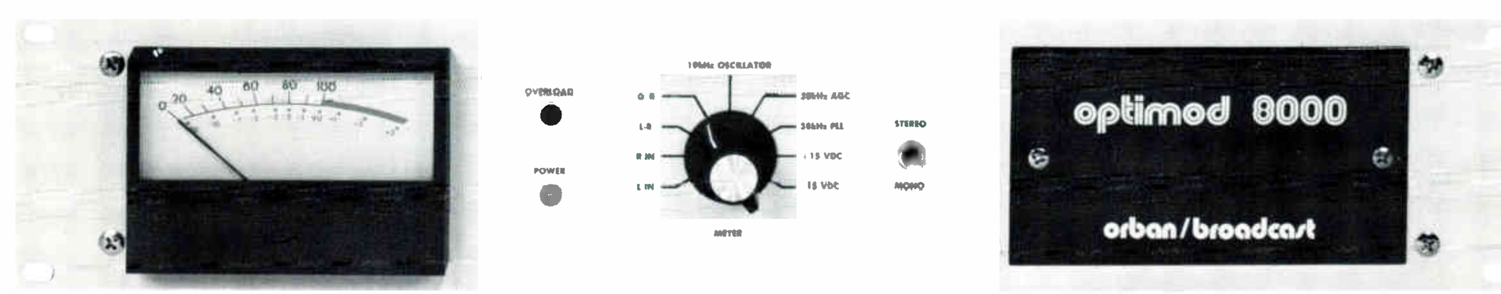
These scope photos compare the 1 kHz square wave response of the OPTIMOD lowpass filter with a conventional stereo generator's lowpass filter. Ringing in the conventional unit forces the steady state level down to 70% modulation to avoid overmodulation – a loudness loss of 3.1 dB compared to OPTIMOD.

A FINAL THOUGHT

OPTIMOD-FM is the ideal audio processing device for any format – hard rock to classical. The basic release time is user-controllable and may be adjusted for a specific format, but the range has been controlled so that objectionable results cannot be obtained. It stands to reason that there should be no such thing as a "rock limiter" or a "classical limiter" – both types of music are perceived by the same hearing mechanism, and if the limiter is well enough matched to the ear, then it will perform ideally with any type of music. Only if the limiter is not well-matched to the ear will different compromises be necessary for different formats.

OPTIMOD-FM has undergone extensive and rigorous development and testing in order to assure a long and reliable operational life without obsolescence. We sincerely hope that the availability of the OPTIMOD-FM will restore the audio quality of the FM medium to the true high fidelity status that was so much a part of Major Armstrong's original vision.

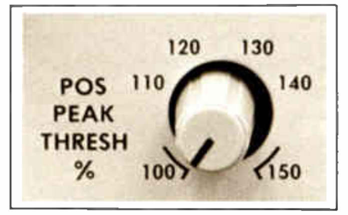
ORBAN/BROADCAST is a brand name for certain products manufactured and sold by the Orban Associates Division, Kurt Orban Co., Inc., San Francisco, California.



Asymmetry And Polarity Followers

A lot of careful listening has convinced us that several of our colleagues in audio processor design and manufacture are right: symmetrical *is* cleaner. Any processor which uses aggressive amounts of clipping for peak limiting will produce only odd-order harmonic and IM products when clipping symmetrically. Asymmetrical clipping produces *both* odd- and even-order products, sounding somewhat brighter, significantly dirtier, and only slightly louder.

The 9100A incorporates a phase scrambler early in the system to make peaks as symmetrical as possible. Operated symmetrically, the 9100A will produce an extremely loud and silky-clean sound, free from midrange "grit" so often associated with the sound of soft-clipping asymmetrical AM processors. You really *can* hear the absence of even-order IM in the midrange! However, for those broadcasters desiring the loudest possible sound, a POSITIVE PEAK LEVEL control permits you to adjust asymmetry to beyond +125% modulation.



The choice is yours, and will depend on format, target audience, and other programming considerations.

Because of the phase scrambler, any natural asymmetry in the input material is eliminated and most efficient use is made of the processing. For this reason, no automatic polarity switching is included (or desirable) in the 9100A. Any asymmetry at the output is artificial and is produced by the processing itself as controlled by the user.

Summary

The 9100A is an exciting development in AM audio processing. For the first time, it permits extremely high loudness to be achieved along with the openness and freedom from processing artifacts heretofore only associated with such FM processors as our OPTIMOD-FM Model 8100A. Simultaneously, its circuitry is substantially less complex than that of our previous Model 9000A, resulting in greater value and higher reliability.

While the 9100A is stereo-ready, the mono station will benefit fully from its use. FM's edge is not as much stereo as it is *quality*. Many now feel that AM has been damaging *itself* by being strident, busy, and over-processed. With the 9100A, programmers and engineers now have a friendly, well-honed tool to create a sound which complements their creative programming: a sound which feels much like FM in its openness, depth, and definition. Early field tests have suggested that ratings will move in the right direction: Fatique is reduced, so more people listen longer.

For many AM broadcasters — particularly those with music formats — the sound of the 9100A may be the key to recapturing audience and ratings.

Order Items

9100A/1 Mono Unit (later easily convertible to stereo with plug-in cards)

9100A/2 Stereo Unit (can readily be used for mono)

RET-17 Optional Lowpass Filter Card needed for certain AM stereo systems, European broadcast, and special adjacent channel problems.

orban

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Specifications

It is impossible to define the listening quality of even the simplest limiter or compressor on the basis of the usual specifications because such specifications cannot adequately describe the crucial *dynamic* processes which occur under program conditions. These dynamic processes are evaluated at the factory and controlled to close tolerances. The measurements require special test fixtures, cannot be readily duplicated by measurements with standard test equipment in the field, and cannot be described in familiar terms.

Certain specifications are therefore presented here to satisfy the engineer that they are reasonable, to help plan the installation, to help make certain comparisons with other processing equipment, and to verify that OPTIMOD AM can readily pass a Proof of Performance. In order to facilitate this, all equalization can be switch-bypassed to enable a Proof to be performed under "flat" conditions as required by the FCC (U.S.A.).

INPUT
Impedance: Greater than 10 K ohms, electronically balanced by means of true instrumentation amplifier. RF suppressed.
Sensitivity: Normal operation may be achieved with nominal line levels of -30dBm or greater. Input sensitivity is controlled by means of INPUT ATTENUATOR control, and also by means of a bypassable 20dB pad before the input amplifier.

INPUT CONDITIONING FILTER
Highpass: -0.5dB @50Hz with rolloff exceeding 18dB/octave below that frequency. Includes deep 25Hz notch for automation cuetones.
Lowpass: Rolls off frequencies above 12kHz at rate exceeding 24dB/octave.
Allpass: Phase scrambler makes peaks more symmetrical to best utilize the capabilities of the processing.

BROADBAND COMPRESSOR
Range of Gain Reduction: 25dB
Compression Ratio: greater than 10:1
Time Constant Mode: SINGLE/MULTI, switch-selectable
Attack Time: approximately 200ms (SINGLE); program-controlled (MULTI)
Release Time: approximately 3dB/second (SINGLE), program-controlled (MULTI)
Total Harmonic Distortion: does not exceed 0.05% at any degree of gain reduction, 50-12,000Hz
Noise: greater than 85dB below output clipping level
Gating: gain will drift slowly to 10dB G/R if input level drops below a user-adjusted threshold
Variable-Gain Element: proprietary class-A VCA

PROGRAM EQUALIZER
Bass: "Quasi-Parametric" second-order peak boost equalizer. "Q" variable from 0.3 to 1.4. TUNING variable from 70 to 110Hz. EQ variable from 0 to +6dB.
High Frequency: Proprietary third-order shelving equalizer (patent pending) matches inverse of "average" (see text) receiver rolloff to a high frequency limit, adjustable from no equalization at all, through any frequency up to 6kHz.
Noise and Distortion are substantially below other elements in the system.

SIX-BAND LIMITER
Filters: 150Hz lowpass; 420Hz bandpass; 700Hz bandpass; 1.6kHz bandpass; 3.7kHz bandpass; 6.2kHz highpass.
Filter Selectivity: 18dB/octave
Filter Topology: parallel.
Filter Combination:
Static: outputs of all filters combine to yield static frequency response ±0.5dB throughout the range of 50-12,000Hz.
Dynamic: Phase interaction between filters under program conditions will not cause audible dips in the frequency response.
Limiters:
Range of Gain Reduction: 25dB.
Attack Time: program controlled; adjusted according to band frequency.
Release Time: program controlled; adjusted according to band frequency.
Total Harmonic Distortion (each limiter): does not exceed 0.1% for any frequency in each limiter's passband with any degree of gain reduction, provided signal is below multiband clipper threshold.
Distortion Cancellation: all clipper-induced distortion in upper four bands cancelled better than 30dB below 1.8kHz. Additional distortion reduction provided as function of frequency in each band.
Noise (each limiter): better than 85dB below VCA output clipping level.
Variable-Gain Elements: proprietary class-A VCA's.

OUTPUT FILTER
Filter Characteristics: 12kHz 5th order elliptical is standard. System guaranteed to meet all requirements of FCC 73.40.a.12 regarding occupied bandwidth for arbitrary adjustments of processing controls and arbitrary program material, provided that the transmitter does not add significant high-frequency harmonic distortion to its spectrum.
Optional Filters: Optional plug-in card contains two phase-corrected 5kHz 30dB/octave filters (for L + R and L − R processing). Also contains a delay network which can be inserted in the L + R path to match the delay of the L − R 5kHz filter if 5kHz bandwidth limitation is desired in the L − R path only. Filters are coupled to the system DAY/NIGHT logic. The card may be strapped in any one of four configurations.

	DAY		NIGHT	
OPTIONS	L + R	L − R	L + R	L − R
1	12kHz	12kHz	5kHz	5kHz
2	12kHz	5kHz	5kHz	5kHz
3	12kHz	5kHz	12kHz	5kHz
4	5kHz	5kHz	5kHz	5kHz

TRANSMITTER EQUALIZER
Low Frequency Tilt Equalizer: Proprietary phase/magnitude compensation introduces adjustable positive-slope tilt to the output waveform to cancel normal negative-slope tilt in older-technology transmitters. Independent control of very-low-frequency compensation available to avoid saturation and non-linear effects in transmitters with limited low frequency power handling capacity.

High Frequency Shelving Equalizer: Adjustable breakpoint shelving equalizer creates controlled undershoots in high frequency transient waveforms to prevent RF envelope overshoot due to excessive "Q's" in transmitters, phasors, and/or antenna systems, or due to poor transient response in audio or modulator stages.

High Frequency Delay Equalizer: Introduces added time delay selectively into the spectrum to compensate for non-linear group delay in the transmitter/antenna system, thru optimizing transient response by creating approximately constant time delay at all frequencies within the audio bandwidth.
Controls: Four separate sets of controls are provided which can be independently adjusted for DAY/TX1, DAY/TX2, NIGHT/TX1, and NIGHT/TX2, and can be remotely switched by momentary application of 6-24V AC/DC between the appropriate terminals on the rear-panel barrier strip. Day/Night and TXI/TX2 status is indicated by pairs of LED's on the front panel. A test point is located behind the control access door. Either a sinewave or squarewave oscillator may be used to drive this test point for TX equalizer alignment.

LINE DRIVER
Output Impedance: 290 ohms, electronically-balanced to ground. RF suppressed by means of third-order non-overshooting EMI filter.
Output Level: will drive greater than +20dBm into 600 ohms.
Configuration (mono): Outputs for two transmitters, each with independent TX EQ and 18-turn screwdriver-adjust OUTPUT ATTENUATOR controls.
Configuration (stereo): Outputs for two transmitters (each transmitter having its own stereo generator), each output with independent TX EQ and 18-turn screwdriver-adjust OUTPUT ATTENUATOR control. Outputs can be strapped for L & R, or L + R and L − R depending on the needs of the subsequent stereo generator.

COMPLETE SYSTEM (PROOF MODE)
(**Note:** PROOF mode requires that all control circuitry for compression and limiting be defeated, and that the program equalizer and TX equalizer be switched OUT, leaving all active circuitry other than the equalizers in-line.)
Frequency Response: better than ±1.0dB, 50-7500Hz (optional 5kHz filters defeated).
Total Harmonic Distortion: less than 0.2% at 100% modulation, 50-7500Hz.
RMS Noise: better than 65dB below 100% modulation, 30-20,000Hz.
Stereo Separation: better than 25dB, 50-10,000Hz; typically 35dB

PHYSICAL
Dimensions: 19"W x 7"H x 12.5"D (4 EIA rack units); (48.3cm x 17.8cm x 31.8cm).
Shipping Weight: 27 lbs. (12 Kg.) Net; 38 lbs. (17 Kg.) Gross
Operating Temperature: 0-50 degrees C (32-122 degrees F).
EMI Environment: Circuitry shielded against EMI from 500kHz - 1 GHz.
Access: Circuitry (except for power supply regulator) on plug-in cards. All circuitry and user setup adjustments available from front panel without removing unit from rack. Control access door is fitted with a lock to prevent unauthorized adjustments.
Power Requirements: 115/230V AC ±15%, 50/60 Hz.
Maximum AC Leakage to Chassis: 0.25mA @115VAC; 0.5mA @230VAC.

WARRANTY
One year, parts and labor. Subject to limitations set forth in our Standard Warranty Agreement. Factory assistance and service available throughout the life of the product.

800020-000-01 8/82

9100A OPTIMOD-AM

MONO OR STEREO

RADICALLY IMPROVED. AND LOUDER, NATURALLY.

OPTIMOD-AM:
An integrated audio processing system for AM radio, including compressor, program equalizer, multiband limiter, clipper, and transmitter equalizer

Performance Highlights

- ☐ Supplies very high average modulation (loudness), exceptional fidelity, uncanny naturalness, and freedom from processing artifacts to yield an FM-like sound from *typical* auto, portable, and table radios.
- ☐ Designed for universal application in domestic or international LW, MW, and SW services; easily adaptable to relevant government standards.
- ☐ Compensates for receiver high frequency rolloff with statistically-derived, adjustable equalizer to extend perceived bandwidth at the receiver without introducing midrange response anomalies or receiver tuning difficulties.
- ☐ New six-band limiter with distributed, distortion-cancelled multiband clipper yields at least 3dB increase in RMS modulation levels compared to old Model 9000A.
- ☐ Consistent output level and equalization texture over a 25dB input level range.
- ☐ Versatile and simple setup controls let you quickly arrive at the sound you really want.
- ☐ Improved transmitter equalizer corrects tilt and ringing in older transmitters and antenna systems for maximum modulation.
- ☐ Outputs for *two* transmitters with independent output level control and full remote control switching for TX1/TX2 and DAY/NIGHT status: i.e., *four* sets of TX EQ adjustments available, all remote-selectable.
- ☐ Sum-and-difference stereo or mono versions available; mono unit fully ready for stereo processing by simply plugging-in additional circuit cards.
- ☐ Orban-quality construction, documentation, support, and service.

Why A New OPTIMOD-AM?

Strangely enough, because of the successful new OPTIMOD-*FM* Model 8100A.

Some of the innovations introduced in it turned out to be highly applicable to AM processing. Specifically, the FM unit introduced a simpler, more economical, and more elegant means of realizing the now-patented distortion-cancelling clipper first introduced in the old 9000A OPTIMOD-AM. And the FM unit featured a distributed crossover with embedded clipper (now patented, too). This concept, extended to six bands and combined with the distortion-cancelling filter, is the key to the higher loudness and astonishingly improved naturalness of the new 9100A OPTIMOD-AM.

A Smarter "Smart Clipper"

In the previous AM unit, the outputs of the six bands in the six-band limiter are combined, fed through a voltage-controlled amplifier, and then applied to a distortion-cancelled clipper. A complex circuit we call the "Smart Clipper"™ controls the gain of the VCA by estimating the amount of *audible distortion* caused by the clipping process, and reducing the VCA gain until such distortion is no longer objectionable.

This, alas, is a *wideband* control process, and must therefore not be over-used if its operation is to be inaudible. In the quest for ever-higher loudness, many Model 9000A users have chosen to operate the "Smart Clipper" with so much gain reduction that its operation is audible. It therefore became clear to us, as designers, that it would be far better to eliminate the need for *any* wideband gain reduction between the output of the multiband section and the final safety clipper. That way, wideband modulation effects would never occur.

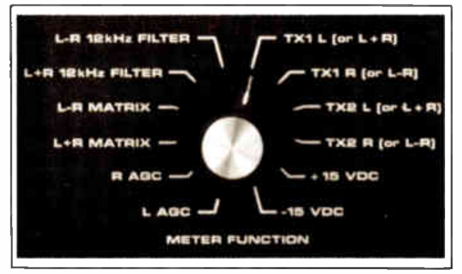
This is more easily said than done. In the previous AM unit, the "Smart Clipper" control circuitry is absolutely necessary to avoid either unacceptable loudness loss or unacceptable distortion on certain program material. The key turned out to be *multiband* clipping combined with distortion cancellation, using our patented techniques in concert with some new developments.

The Bottom Line: FM-Like Performance

The exciting result of such processing is that the combined output of the bands can be fed to a safety clipper *without* interspersing wideband gain control — provided that almost 100 internal parameters are correctly "tuned" in the design process! However, the payoff is worth it: a dramatically open, effortless, multiband sound with literally *no* audible processing on virtually any radio likely to be in the hands of your audience. A sound which is "FM-like" not only in terms of frequency response, but also in terms of "punch", "depth", "openness", and "definition". And a sound which, on a true RMS meter, averages about 3dB higher than the previous unit for the same peak modulation. In short a *loud* yet unbelievably natural sound which, we think, stands the best chance yet of winning back an audience becoming more and more attracted to FM.

AM Stereo

The 9100A is available as a stereo processor, or as a stereo-convertible mono processor. Stereo conversion is achieved simply by plugging in circuit cards — no "accessory chassis" is required except for the stereo generator.



Processing occurs in the "sum-and-difference" mode, which is most appropriate for AM stereo because the AM modulation component represents the sum (L + R) of the channels to assure compatibility with mono receivers. Internal straps determine if the output is to be in L + R and L–R mode, or in L and R mode, yielding complete versatility in matching it to the stereo generator.

The standard bandwidth of the 9100A is limited to 12kHz by means of highly selective filters, enabling it to comply with the occupied bandwidth requirements of FCC 73.40.a.12 with arbitrary program material and processing adjustments. An optional plug-in filter card, provided at extra cost, (which fully interfaces with the DAY/NIGHT remote control) permits you to limit the bandwidth of the L-R channel to 5kHz, controlling potential IM distortion which can be introduced by high-energy high frequency information in some of the AM stereo systems.

This card also has straps which permit realization of virtually any combination of 12kHz and 5kHz bandwidths in the sum and/or difference channels, in DAY and/or NIGHT modes, enabling broadcasters to limit mono or stereo bandwidth to 5kHz at night (to control interference to other stations), or to operate at 5kHz at all times to meet EBU or other international specifications.

Stereo or Quality: What Really Attracts Listeners?

Despite the fact that the 9100A is well-equipped for stereo, we suspect that this may not be as important as some people think.

A very popular local FM station lost its separation — but not its pilot, so listeners' stereo lights did not go out. The result? In three days, *one* listener called to complain — and only because he noticed the problem on the vector scope in his expensive tuner. None of the staff noticed anything awry. The station's contract engineer finally noticed the problem and quietly fixed it.

So what? Compared to attaining FM-like audio quality, AM stereo *sound* (as opposed to a little light on someone's radio saying "stereo") just might be a secondary consideration for achieving audience ratings. The 9100A is unique in its ability to make you *sound good*, whether in mono or stereo. And we think that *sounding good* is what is going to bring listeners back from the FM band. Judge for yourself!

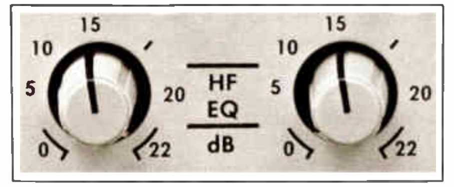
The 9100A: Designed For The Real World

Operator Gain-Riding: A classic problem of multiband compressors is their sensitivity to input levels. Incorrect operator gain riding can change frequency balances and equalization textures in a disturbing way.

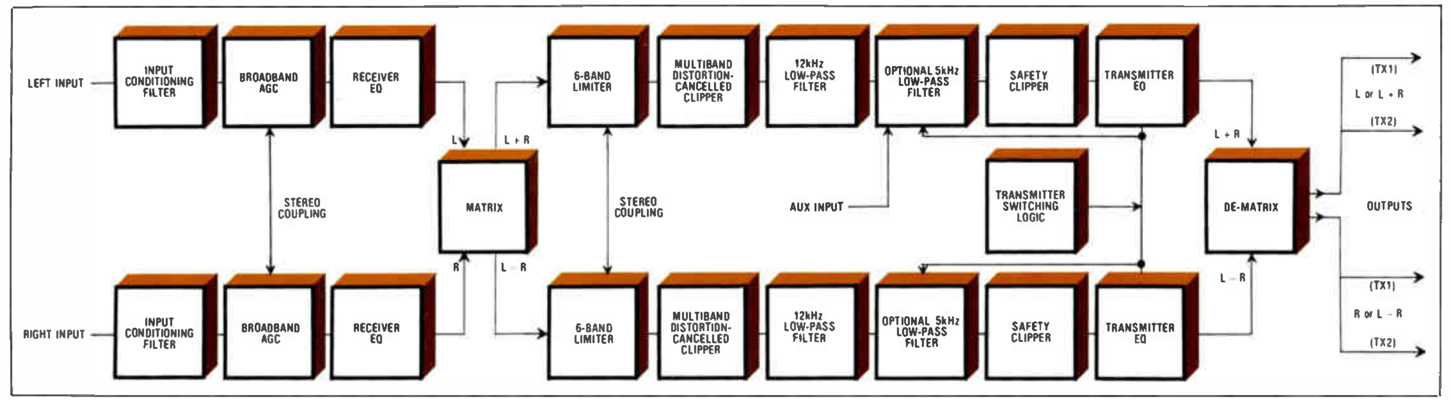
Because of this, the 9100A has a newly-designed AGC amplifier ahead of its six-band limiter. It is designed to do as little as possible to the sound except to slowly gain-ride over a 25dB range. Accordingly, despite its being a wideband device, it does not make nasty wideband sounds. And it relieves the operator of the requirement of reading a complicated array of flashing LED's just to determine if the processor is being driven correctly!

Receiver Equalization: We carefully measured the frequency response of fifteen of the most common real-world AM radios. We then averaged the curves, and did a statistical analysis to make sure that our procedure meant something. Finally, we mathematically synthesized an inverse (preemphasis) curve which can correctly equalize the radios flat up to a -3dB point of 6kHz. We took this curve and designed an equalizer circuit which could create it, or any part of it. So broadcasters who don't wish to equalize out to the full 6kHz (which requires *lots* of high frequency boost) can equalize out to 5kHz, 4kHz, 3kHz, or whatever — all without introducing midrange coloration, as offsetting the top band on a conventional triband processor inevitably does when you try to equalize for radio rolloffs.

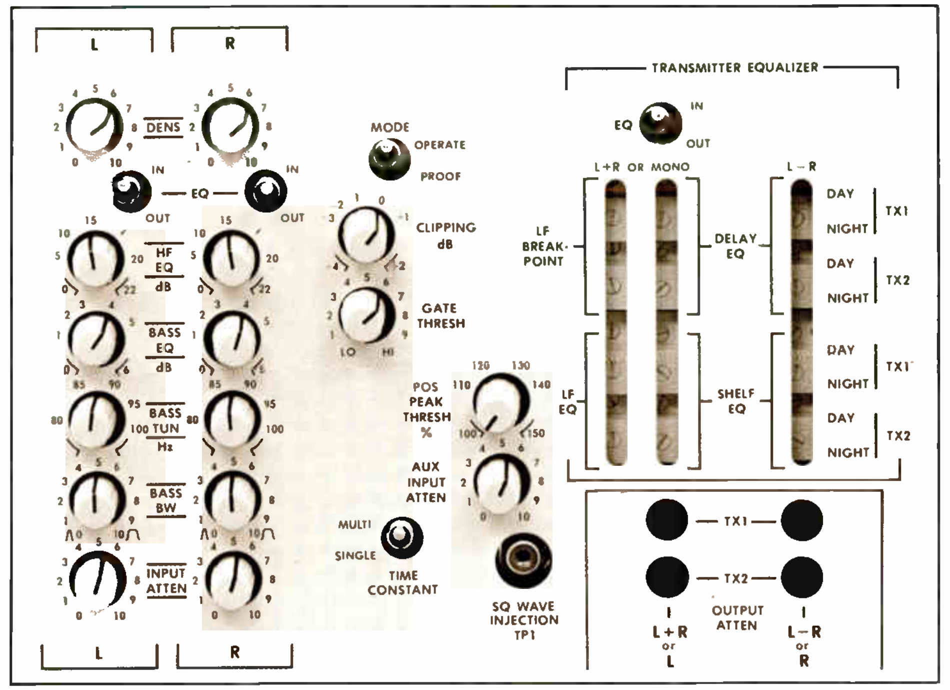
The result is a *one-knob* high frequency equalizer that can produce *nothing but correct equalization curves!* Much easier to use than the three-knob H-F equalizer on our previous AM processor. And much more accurate, too!



We then complemented this unique H-F equalizer with a versatile *parametric* bass equalizer which can tune bass response to your format and target audience.



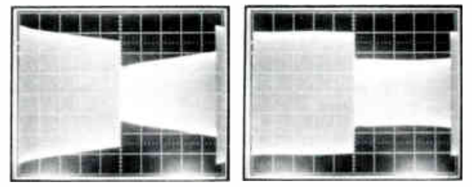
SYSTEM BLOCK DIAGRAM
(STEREO CONFIGURATION)



SUBPANEL SETUP CONTROLS

Finally, to keep D.J.'s happy, we synthesized a passive monitor rolloff filter with an H-F ROLLOFF control to match it to the processor H-F EQ setting you've chosen. This filter can be inserted between your mod monitor output and your monitor amplifier input to produce a big, high-fidelity sound even on large studio monitor speakers. The filter is included standard with every 9100A.

Transmitter Equalization: Not everyone is fortunate enough to operate a state-of-the-art transmitter with its negligible tilt and overshoot. Older plate-modulated transmitters can have enough tilt and/or overshoot to substantially compromise the accuracy with which they can reproduce a highly-processed signal like the output of the 9100A. And even state-of-the-art transmitters can ring into high "Q" antenna systems. Often, average levels must be reduced to accomodate the peak level increases introduced by such inaccuracies.

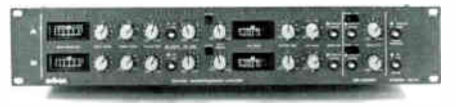


WITHOUT TILT CORRECTION / WITH TILT CORRECTION

LOW FREQUENCY SQUARE WAVE

Our previous processor was equipped with a transmitter equalizer to "tune out" tilt and overshoot — this equalizer proved very useful in practice. The 9100A includes an augmented version of this circuit which adds yet another refinement to the tilt equalizer. This allows you to control the amount of very-low-frequency correction introduced, permitting you to match the equalization to the transmitter as accurately as possible while simultaneously avoiding saturation of the modulation transformer (or other circuitry) in transmitters which cannot "take" the full correction.

Protecting STL's: Unlike our current OPTIMOD-FM and -TV, the 9100A system cannot be split into separate studio and transmitter sections to provide compression before the STL. To accomodate those who need overload protection for their STL's, we recommend the use of an Orban 422A (mono) or 424A (stereo) compressor/limiter/de-esser at the studio side of the STL.



THE 424A

These economical, high-quality units are well-matched to the 9100A, costing no more than a "9100A Studio Accessory Chassis" otherwise would.

Auxiliary Input: A separate input is provided after all processing except the safety clipper and transmitter equalizer. Located in the L + R (or mono) channel, this provides a convenient input for injecting EBS tones, subsonic telemetry, and the like. It also provides a convenient point into which the output of a separate voice-processing audio chain can be mixed for those stations desiring independent voice/music processing.

Packaging and Maintenance: The 9100A is packaged in the form of plug-in cards, significantly facilitating maintenance. The cards plug into a rugged chassis with effective, field-proven RFI shielding, making the system operable in almost any EMI environment without difficulty.

Loaner cards are available from the factory, and suspected field defects are preferably verified by replacing a questionable card with one known to be good. Those wishing to do in-house maintenance of the 9100A system will be greatly aided by an outstanding complete operation and maintenance manual which contains detailed circuit descriptions, standard curves, and other helpful data.

Applications

The 424A can be applied anywhere that current compressors, limiters, and/or de-essers are used, since its versatility does not limit it to a single "sound," but instead lets it operate as a smooth, gentle compressor, a peak limiter, a de-esser, or any combination of these — all with a singular absence of undesirable artifacts.

This means that the unit can be used in recording studios, in broadcast production studios, ahead of broadcast studio-transmitter links, in sound reinforcement, and in video production and sweetening. It is an ideal all-in-one vocal processor, combining the necessary AGC and de-esser functions. It also shines when smoothly handling **mixed** program material — making it excellent for preparing cassette duplicating masters, or protecting tape recorders and cart machines from overload in tight time-pressure situations in broadcast and recording. Simultaneously, its versatile, wide-range setup controls make it a natural for processing single instrumental or vocal tracks in studios. Exploitation of the VCA clipping feature can result in substantially more natural peak limiting than most simple "limiters" can provide, resulting in improved performance when protecting broadcast STL's or power amplifiers in sound reinforcement systems.

Summary

The 424A "Studio Optimod" is the answer to many engineers' dreams. It combines a compressor, limiter, and de-esser in a most versatile way. Because its controls interact in a carefully human-engineered manner, it is easy and graceful to operate. Yet full flexibility is there to get the sound just right.

The professional audio and broadcast world has lived happily with "old favorite" limiter/compressors for a long time. If you examine the features, sound, performance, and price of our new "Studio Optimod" Model 422A/424A, we think you will agree that it is the new standard in dynamic range control. But we don't expect you to take our word for it. The proof is in the **listening.** We feel confident that once you A/B our unit against any of your current favorites, you will find a place for it in your rack. It's that good. A truly superior device, at the right price, at the right time.

Rest assured that with your new 422A/424A, you will continue to receive all the other things that you've come to expect from Orban products over the years — quality construction, comprehensive operating and service manuals, and unequalled customer service.

SPECIFICATIONS

INPUT
Impedance: greater than 10 k ohms active balanced; RF suppressed
LEVEL: −15dBm produces 10dB gain reduction with ATTACK TIME control centered, INPUT ATTEN control fully CW, and RATIO control at infinity-to-one

OUTPUT
Impedance: approximately 100 ohms, electronically balanced to ground; RF suppressed
Level: +4dBm nominal; absolute peak overload occurs at +26dBm

FREQUENCY RESPONSE
±0.25dB 20-20,000 Hz below limiting and de-esser thresholds

COMPRESSOR/LIMITER SECTION
Attack Time: manually adjustable in approximate range of 500us to 200ms; automatically scaled by program content
Release Time: adjustable in approximate range of 0.8dB/sec to 20dB/sec; automatically scaled by program content. Switch-selectable LINEAR and EXPONENTIAL release shapes.
Compression Ratio: adjustable from 2:1 to infinity-to-one at threshold. Lower ratios automatically increase beyond threshold.
Range Of Gain Reduction: 25dB
Tracking Of Multiple Channels: ±0.5dB
Total Harmonic Distortion (ATTACK and RELEASE TIME controls centered; infinite RATIO; 15dB gain reduction): less than 0.03% @1kHz. Typically 0.11% @20Hz; 0.02% @100Hz; 0.01% @1kHz; 0.04% @10kHz.
SMPTE IM Distortion (controls set as above; 60/7000Hz 4:1; 15dB gain reduction): typically 0.05%

DE-ESSER SECTION
Attack Time: approximately 1 ms
Release Time: approximately 30 ms
Harmonic Distortion: less than 0.05% THD introduced by de-essing action @10kHz
Available Gain Reduction: greater than 25dB

SYSTEM NOISE
RMS noise in 20-20kHz bandwidth better than 85dB below output clipping threshold for any degree of gain reduction; 90dB typical.

OPERATING CONTROLS
Compressor/Limiter
INPUT ATTENUATOR
COMPRESSION RATIO
ATTACK TIME
RELEASE TIME
RELEASE SHAPE
GATE THRESHOLD
OUTPUT TRIM
IDLE GAIN
COMPRESSOR/LIMITER OPERATE/DEFEAT
OUTPUT LEVEL (REAR PANEL)
De-Esser
THRESHOLD
DE-ESSER OPERATE/DEFEAT
General
STEREO COUPLING (424A only)
POWER ON/OFF

INDICATORS
Compressor/Limiter
GAIN REDUCTION METER
GATED LED
VCA LEVEL METER
De-Esser
NORMAL De-essing LED
HEAVY De-essing LED

Power Requirement
115/230 VAC ±10%; 50-60Hz. U-ground power cord attached; RF suppressed

Dimensions
19" (48.3cm) wide x 3.5" (8.9cm/2 units) high x 10" (25.4cm) deep

Operating Temperature
0-45 degrees C

Warranty
One year, parts and labor. Subject to limitations set forth in our Standard Warranty Agreement.

orban

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The Orban 424A Gated Compressor/Limiter/De-Esser

The Studio Optimod

Performance Highlights

- ☐ Intuitive and natural operation
- ☐ Adjustable attack time, release time, and compression ratio permit extremely natural processing or special effects
- ☐ Selectable linear (general purpose) or exponential (special purpose) release time characteristics
- ☐ Defeatable gate with adjustable threshold causes gain to move slowly toward user-adjustable value during pauses, preventing noise rush-up, pumping, or breathing
- ☐ Major controls interact to speed setup by keeping output levels relatively constant as controls are adjusted
- ☐ **Separate** compressor/limiter and de-esser control loops, each with optimized, program-controlled parameters
- ☐ Better than 25dB de-ess gain reduction available **in addition to** 25dB compressor/limiter gain reduction
- ☐ True peak-reading VCA LEVEL meter
- ☐ True peak-reading GAIN REDUCTION meter
- ☐ De-esser characteristics similar to highly-accepted Orban dedicated de-essers
- ☐ Low-distortion operation achieved using clean class-A VCA and distortion-cancelling control circuitry

Plus

- ☐ Rugged all-metal 19" rack mount package for ruggedness, roadworthiness, RFI shielding. Industrial-grade parts and construction
- ☐ Highly cost-effective; available in mono (model 422A) or stereo (model 424A)
- ☐ Multiple channels can be connected to track together
- ☐ Extensive RFI supression
- ☐ Balanced inputs and outputs, and 115/230V 50-60Hz power supply standard

THE 422A IS A SINGLE-CHANNEL VERSION OF THIS PRODUCT

The Model 422A/424A:

A multi-function Compressor/Limiter/De-Esser featuring exceptional versatility and ease of operation

Preface

We're about to say a lot about our 424A: we're proud of it. But no long essay can describe the bottom line — what it sounds like, and how it feels to use it. Comparisons of specifications and descriptions of new control techniques cannot describe the elusive and magical relationship between engineering and hearing.

This brochure should answer questions you might have about why the 424A is such a good-sounding and easy to use product. But when you get right down to it, the best way to appreciate its advanced design is to A/B it against your current favorite. Using it and listening to it are what really count!

The Orban 424A: It Had To Be Better, Or We Wouldn't Have Bothered

There are a lot of production limiters out there. Old favorites. Pretenders to the throne. The competition is fierce, and the market fragmented. So when Orban set out to design a significant new production limiter, we knew it had to be superior.

Fortunately, when we undertook the 424A R&D project, we had seven years of experience behind us. Enough to capture the Number One slot in the broadcast signal-processing market. Ask any FM broadcast engineer what the industry-standard limiter is, and he's likely to tell you it's our OPTIMOD-FM.

We've developed the technology — the **magic** which makes a compressor/limiter sound right, feel right, and operate quickly and intuitively. That's important in today's economic climate, where audio professionals demand fast, superlative results in recording studios, in broadcast production studios, and on the road doing demanding sound reinforcement work.

The result of our research and experience is the model 424A — a Gated Compressor/Limiter/De-esser with versatile features for production, and with a natural, transparent sound that has to be heard to be appreciated.

Our goal was to build a device which would produce the desired sound upon adjustment of a **minimum** number of controls. And, readjustment of one control should not require the corrective readjustment of others. We achieved this goal by making the ATTACK TIME, RELEASE TIME, and COMPRESSION RATIO controls interact with each other and with the threshold of limiting. For example, slower attack times permit more overshoots, so the threshold of limiting is automatically lowered to compensate. The result: you can concentrate on getting the **sound;** the mechanics largely take care of themselves.

The Un-Trendy Limiter

Most of the advertising buzzwords applied to AGC units are irrelevant to the essential listening qualities of the devices. Simultaneously, many of the truly important issues of AGC design (many of which are quite subtle and not easily reduced to buzzwords) seem to be unknown to, or ignored by, others.

The 424A is a very un-trendy device, in that it uses feedback control, an averaging detector, and a conventional "hard knee" static compression curve. Why?

For starters, because the type of detector (whether true-RMS, "linear-integration" averaging, or whatever) is essentially irrelevant, since far more variation results from simply changing the attack time than from changing the detector type. Similarly, the desirability of a consistent output as read on a VU meter is questionable, since VU meter readings have only the most casual correlation to psychoacoustical loudness.

Feedback control circuitry has been accused of hiding the vices of inferior VCA's while introducing instability and high amounts of distortion. Our VCA has nothing to hide, our loop is totally stable, and our measured THD is significantly better than most others on the market — feedback or feedforward. We use feedback because our ears tell us that, properly implemented, it creates a control loop dynamic response which simply **sounds better.**

The "soft knee" compression ratio characteristic promoted by others is a way of making a compressor sound innocuous by "sneaking up" upon high ratios over the course of many dB. Low ratios always sound more graceful than high ratios. In a "soft-knee" compressor, mostly low ratios are used. No wonder the sound of the compressor is improved! Alas, the price exacted is lost loudness and inconsistent level control. This may be fine for certain applications, but not if you're trying to persuade a wide dynamic range vocal to cut through a heavy instrumental backing, or if you're trying to make anything audible at all times.

What then, **does** make a good-sounding compressor/limiter? Primarily, the dynamic response of the control loop. In the 424A, attack and release times are always "automatic" (i.e., program-controlled); the ATTACK TIME and RELEASE TIME controls merely scale the processes faster or slower without giving up the advantages of the program control.

If a slower attack time is chosen, an automatic gain-riding (AGC) function is achieved. Faster attack times introduce more peak limiting. So, depending on attack time, the 424A can serve as a compressor, limiter, or both simultaneously.

What Makes The 424A So Special?

The Orban 424A has a real compression ratio control for those few times when you want to maintain some **real** dynamic range. However, you will probably be astonished at the "openness" and **apparent** dynamic range available even at the "infinite" ratio. You may be even more astonished when you discover the apparent loudness increase achieved without the usual side effects. Virtually any competent compressor can sound natural if it is quiet enough (i.e., if it doesn't actually work very hard). **The real magic is sounding loud and natural simultaneously, as the 424A does.**

After you've lived with a 424A for a while, we suspect that the dust will build up on the ratio control as you realize that, finally, here is a compressor which doesn't have to be cranked back to a 2:1 or 4:1 ratio (whether manually, by choice, or involuntarily, by means of a "soft knee") to sound good! For those of you who won't give up the "soft knee" no matter what, we've hedged our bets: you'll be pleased to know that at the lower settings of the 424A RATIO control, the ratio increases as more gain reduction is used.

There is also control over the **shape** of the release characteristic. Ordinarily, the release characteristic is "linear." (Recovery, in the absence of program material, proceeds at a constant number of dB per second.)

A switch-selectable "exponential" release shape is also provided for special applications. This forces the release to start slowly and to increase in speed as it proceeds.

While it sounds substantially less natural than "linear" on most program material, it can be useful when gain-riding wide-dynamic range single tracks (like vocalists) where the "open" sound of a slow release time is desired, yet quick gain-riding is necessary to make levels more consistent.

The IDLE GAIN control is a unique and unusually useful new feature. This control sets the gain of the compressor when it is in the "idle" mode. (When the compressor is manually defeated by operation of the DEFEAT switch, or, when it is gated by low-level audio or silence.)

In either case, the gain will move smoothly toward a value specified by the setting of the IDLE GAIN control — quickly after manual compressor defeat, and more slowly under gated conditions.

If the IDLE GAIN control is set close to the average gain reduction, then, upon resumption of ordinary compression, there will be minimum gain change in the VCA, and therefore minimum audible side effects will occur. The feature is extremely useful in preventing noise or tape hiss from being pumped up, and in facilitating smooth manual switching of the COMPRESSOR OPERATE/DEFEAT switch during program. Its effect is usually substantially smoother than that of a conventional noise gate.

The OUTPUT TRIM control can be used to force some peak clipping in the VCA in applications requiring tight control of peak levels (like the protection of a broadcast STL), thus controlling fast peaks without need for extremely fast attack times (which almost invariably cause more audible degradation than a modest amount of overshoot clipping). Conveniently, the peak-reading VCA LEVEL meter not only allows you to adjust the 424A to **clip if you want it to,** but also makes it easy to **avoid clipping entirely** if that is your goal.

About Distortion And VCA's

In any compressor/limiter, the static nonlinearities of the VCA are ordinarily totally overshadowed by the dynamic distortions caused by literally intermodulating the input signal by a rapidly varying control voltage. Sometimes such distortion is heard as additional unwanted spectral components, such as traditional harmonic and IM distortion. At other times, it is perceived as unnatural modulations of the signal peculiar to AGC devices such as "pumping," "hole-punching," "shivering," and a whole bunch more which no one has bothered to name, but which most musically-sensitive people can easily hear!

In the Orban 424A, nonlinear control voltage smoothing results in singularly favorable **dynamic** distortion properties. Our proprietary VCA complements this low dynamic distortion by slewing quickly, having "soft" low-order static distortion which is well below the threshold of audibility (due, in part, to class-A operation), and having noise performance which does not deteriorate as the amount of gain reduction increases.

The Final Coup: A Full-Function De-Esser

Finally, our de-esser. The VCA used in the 424A has **two** gain-control inputs. This means that we can add a **no-compromise** de-ess function which is essentially **independent** of the compressor/limiter. As we have proven in our dedicated de-essers, the optimum attack times and release times are quite different for the compressor/limiter and de-esser functions. In the 424A, each function is independently optimized.

The de-esser section of the 424A sounds about the same as our popular dedicated de-essers. It controls sibilance levels by quick broadband gain reduction when excessive "ess" energy is detected. In this way, any low-frequency IM distortion is reduced along with the "ess," and coloration of the "ess" sounds does not occur. One extra benefit of the de-esser section of the 424A is that it can effectively de-ess sibilant vocals which have already been **mixed** with other program, and, in this application, effectively acts more like a high-frequency limiter.

The only essential difference between the 424A de-esser and the Orban dedicated de-essers is that the 424A lacks the dedicated de-essers' ability to provide constant de-essing regardless of input level. In the 424A, this does not create a problem since it was assumed that the input to the de-esser section would be compressed by the 424A's compressor/limiter and would therefore be at a constant level. LED's indicating NORMAL/HEAVY de-essing allow the SENSITIVITY control of the de-esser to be readily adjusted.

Mono, Stereo, Or Dual-Channel

A STEREO COUPLING switch allows you to operate the two channels of the 424A independently or in stereo. In stereo, the channels will typically track within ±0.5dB. Rear-panel coupling terminals allow tracking an unlimited number of channels together.

The defeatable silence gate (controlled by the front-panel GATE THRESHOLD control) avoids noise rush-up during near-silent passages by radically slowing the release process, thereby forcing the VCA gain to freeze and then to move very slowly toward the average of the last 30 seconds of gain reduction. (This is an automatic "idle gain" function, similar to the manually set IDLE GAIN on our 424A Gated Compressor/Limiter/De-Esser.)

VCA noise in the gain reduction circuitry (typically −90dB below the VCA output clipping point, 20-20kHz unweighted) stays virtually constant as the VCA gain changes. This results in very low noise under real-life conditions. The VCA used is a proprietary Orban Class-A VCA with very low static distortion.

Summary

The Orban Model 464A Co-Operator provides an unprecedented combination of versatility, audio quality, and ease-of-use. Like any such device, specifications and brochures can't really describe what it sounds like or how it feels to use it. We invite you to try the Co-Operator for yourself. We think you will find it to be an indispensible assistant.

Other Orban Level Control Devices

Model 422A (mono)/ 424A (stereo) Gated Compressor/ Limiter/De-Esser: A very smooth-sounding, full-featured, gated compressor/limiter/de-esser. Provides manual control of gate threshold, attack time, release time, compression ratio, idle gain, and de-essing threshold. The 422A/424A is the best choice where the need for additional versatility justifies the time and skill necessary to set the many manual controls, and where HF limiting is not required.

Model 412A (mono)/ 414A (stereo) Compressor/Limiter: A low-cost version of the 422A/424A with an added threshold control (but without gating or de-esser). Ideal for sound reinforcement applications.

Specifications

Performance

INPUT

Impedance: >10K ohms, active balanced, EMI-suppressed.

Operating level: Usable with nominal levels from −10dBm to +8dBm.

OUTPUT

Impedance: 93 ohms, electronically balanced and floating to simulate true transformer output. Minimum load impedance is 600 ohms. Output can be unbalanced by grounding one output terminal.

Level: Front-panel controls permit use with −10dBm to +8dBm systems. Output clipping level is >+20dBm into 600 ohms.

SYSTEM

Frequency response (20–20,000Hz): ±0.25dB below leveler, compressor, and high-frequency limiter thresholds.

RMS noise (20–20,000Hz): >85dB (90dB typical) below output clipping threshold with high-frequency limiter strapped for flat output.

Interchannel crosstalk: Better than −60dB at 15kHz (−67dB typical), falling at 6dB/octave below 15kHz.

Circuitry

LEVELER/COMPRESSOR

Attack time: Approximately 200ms for leveler, 5ms for compressor; program-dependent.

Release time: Adjustable between approximately 1dB/sec and 5dB/sec; program-dependent. Below 10dB gain reduction, either a constant dB/sec release rate (HARD RELEASE SHAPE) or an exponentially declining rate (SOFT RELEASE SHAPE) may be selected.

Compression ratio: >20:1 (static); program-dependent (dynamic).

Range of gain reduction: 25dB.

Interchannel tracking: ±0.5dB (with MODE button set to STEREO).

Total harmonic distortion: <0.05% at 1kHz (with RELEASE TIME controls centered and 15dB gain reduction). Typically <0.1% at 20Hz, <0.03% from 100–2,000Hz, <0.05% from 2,000–10,000Hz, and <0.1% from 10,000–20,000Hz.

SMPTE intermodulation distortion: <0.05% (60/7,000Hz 4:1 with 15dB gain reduction).

Gain reduction element: Class-A proprietary VCA.

HIGH-FREQUENCY LIMITER

Pre-emphasis: Six switch-selectable 6dB/octave pre-emphasis curves: 25, 37.5, 50, 75, 100, and 150 μs. Can be strapped for flat or pre-emphasized output. A defeatable peak clipper can enforce an absolute peak ceiling on the (pre-emphasized) output of the HF limiter.

Response: The high-frequency limiting threshold and attack time have been set so that no audible distortion is produced with dynamic program material that has been processed by the leveler/compressor and peak clipper. Because these settings have taken into account the peak-to-average ratio of the leveler/compressor's output, it is not possible to specify the high-frequency limiter's response to test tones with simple, meaningful numbers.

Total harmonic distortion: The high-frequency limiter/clipper will add no more than 0.02% THD to sinewave test tones that have been processed by the leveler/compressor.

Release time: Approximately 30ms, program-dependent.

Interchannel coupling: Each channel's high-frequency limiter operates independently at all times (the fast release times do not disturb the stability of the stereo image).

Gain reduction element: Junction FET.

HF limiting curve: Shelving, 6dB/ octave.

CONTROLS & METERS

Buttons: "Wink-eye" type—end of buttons turn white when pressed in.

Meters: Four 10-segment LED bargraph displays show gain reduction and peak output level for each channel.

Indicators: Four LEDs light to show operation of gating or high-frequency limiting.

Installation

Power requirements: 115/230 volts AC ±10%, 50–60Hz, 16VA. Three-wire "U-Ground" power cord (to USA standards) attached. EMI-suppressed.

Dimensions: 19.0″ (48.3cm) wide, 9.625″ (24.5cm) deep, 1.75″ (4.5cm) high.

Operating temperature range: 32–113°F (0–45°C)

Input/Output connections: Barrier strip (#5 screws) in parallel with ¼″ balanced phone jacks. Chassis is punched to accept optional XLR connectors.

Warranty: One year, parts and labor. Subject to the limitations set forth in Orban's Standard Warranty Agreement.

orban

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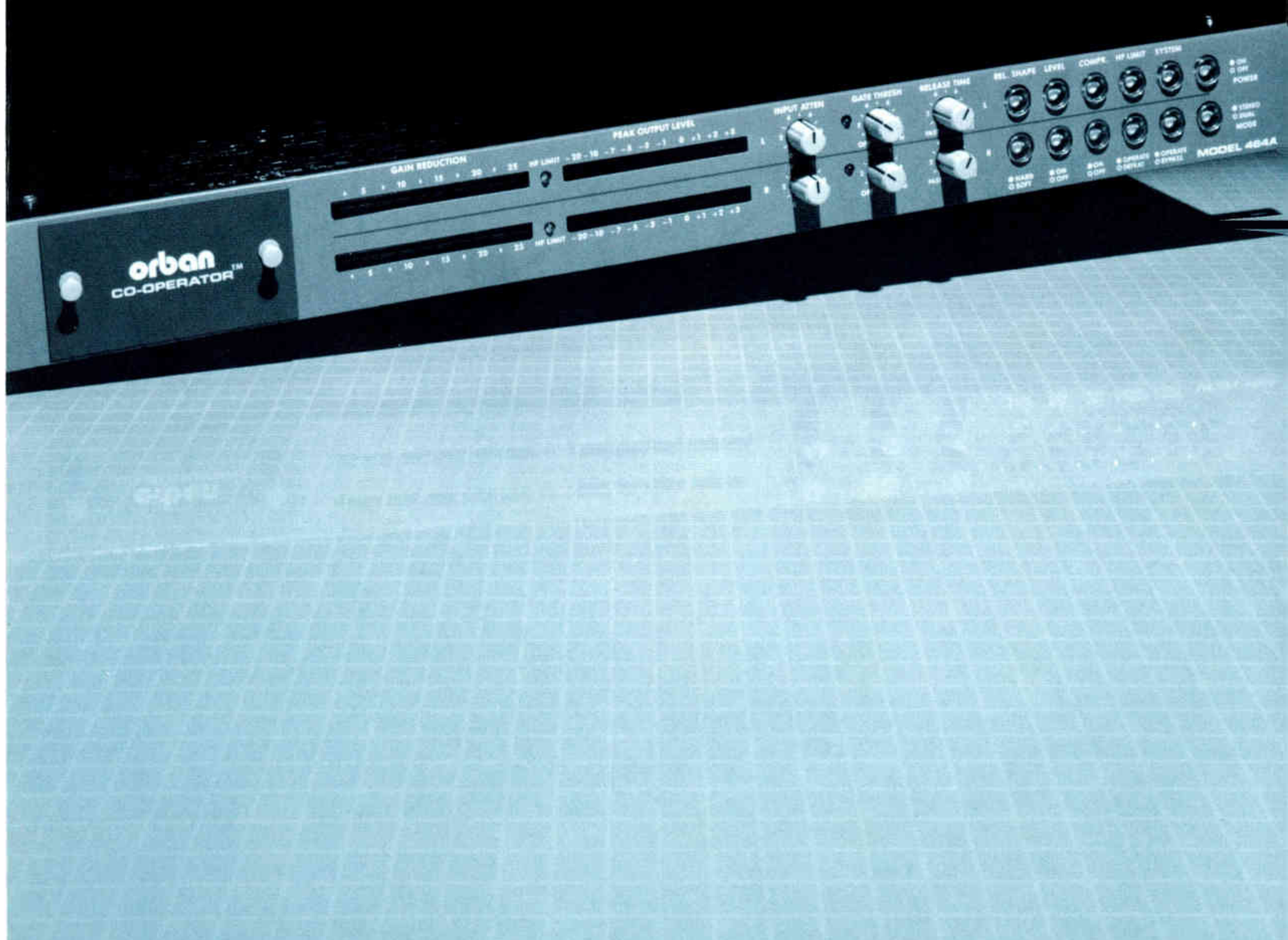
The Orban 464A Co-Operator™

An Integrated, Easy-to-Use
Gated Stereo Leveler / Compressor / HF Limiter / Peak Clipper



Features:

- ☐ A smooth leveler function provides transparent gain riding—without long-term distortion-producing overshoots.
- ☐ A faster "compression" function can be switched in to provide additional transient overshoot protection for material with abrupt level changes or unusually high peak-to-average ratios. The compression function is available without leveling for applications that require safety limiting only.
- ☐ Release time and release shape are adjustable to optimize processing for music or voice.
- ☐ A defeatable "silence gate" prevents noise rush-up, holes, pumping, and breathing by inhibiting sudden gain increases once the signal level falls below a preset threshold (during pauses or low-level program material).
- ☐ Six switchable HF limiter pre-emphasis/de-emphasis curves (25 to 150µs) allow the HF limiting to react optimally for the medium or device being protected and guard against excessive sibilance.
- ☐ A defeatable clipper follows the HF limiter for absolute peak protection.
- ☐ Switchable for stereo-tracking or independent two-channel operation.
- ☐ The least-used controls are concealed behind a security panel to avoid confusing non-technical operators, and to permit tamper-proof calibration.
- ☐ Two LED bargraphs per channel simultaneously display gain reduction and peak output level.
- ☐ Output level meter can be calibrated to match the overload point of the device being driven.
- ☐ Balanced, floating inputs and outputs are EMI-suppressed.
- ☐ 25dB gain reduction range is achieved with a low-distortion, Class-A VCA.
- ☐ Two channels in a space-saving, rugged, all-metal, 1¾" package.
- ☐ Hard-wired bypass switch is included.

Your Assistant Operator

The Orban Co-Operator is a friendly, automatic "assistant operator" that smoothly and unobtrusively rides gain and limits peaks.

After audio quality, ease of use was the highest priority in the development of the Co-Operator. The operator need only be concerned with three controls:

Input Attenuator: Determines the amount of gain reduction.

Gate Threshold: Determines the level below which ordinary automatic gain control (AGC) action is "frozen" (to prevent noise rush-up during pauses and low-level program).

Release Time: Speeds or slows the program-controlled release time to match the processing to the "flavor" of the audio being processed.

The operator can also select the gain reduction recovery rate with the RELEASE SHAPE control. The constant dB/second HARD recovery rate is intended for use with single tracks and live voice; the SOFT rate (release slows down as gain increases) works well with mixed program material.

Push-button selection of the slow AGC LEVEL function (200ms attack) and/or the faster attack COMPRESSION function adapts the Co-Operator to a wide range of level control chores.

The defeatable high-frequency limiter can be used to prevent pre-emphasized material from overloading tape recorders, disk cutters, high-frequency drivers in sound systems, broadcast STLs, FM SCAs, and cassette masters which have excessive high-frequency energy. Controlling high-frequency energy permits average recording or transmission levels to be increased, yielding improved signal-to-noise ratios.

The Co-Operator also offers defeatable peak clippers which follow the high-frequency limiters. The clippers are useful when the device following the unit has an absolute "brick-wall" peak overload point (such as broadcast transmitters, digital tape, and power amplifiers). Typically, a peak clipper produces fewer audible processing-induced artifacts than a fast-attack peak limiter and is, therefore, best-suited for absolute peak protection chores.

As a result of Orban's years of research and practical experience in the broadcast signal processing field, the Co-Operator has outstandingly transparent audio performance. This performance is achieved through the use of finely-tuned control loops to eliminate the dynamic distortions that are the downfall of most conventional compressors and limiters, and through the use of a clean, Class-A proprietary VCA to ensure negligible static distortion and noise.

Easily interpreted, *independent* LED bargraphs display gain reduction and peak output levels for each channel.

Versatility Without Confusion

The Co-Operator offers unprecedented flexibility for the effective and foolproof control of levels in a wide range of professional applications. To ensure ease of use and to avoid confusing the less technical operator, set-up controls and jumpers are concealed.

The controls which calibrate the OUTPUT LEVEL meters and those which determine the actual output level are behind a pop-off door on the front panel, as are the switches which determine the pre-emphasis curves of the high-frequency limiters. The clippers, which follow the HF limiter, are strapped in or out with a jumper on the circuit board. Thus, once the Co-Operator has been set up by the installing engineer, these controls are safely out of sight. (Where the Co-Operator protects an STL, SCA, or transmitter, an optional Security Cover can prevent unauthorized adjustment of *any* control.)

Applications

Recording Studios: With the RELEASE SHAPE control set to SOFT, the outstandingly transparent Co-Operator is ideally suited for subtly reducing the dynamic range of an entire mix. The HARD setting provides effective gain riding on single tracks, increasing "punch" and intelligibility while retaining the basic feel of the performance and the apparent dynamic range of the voice or instrument. The RELEASE TIME control (a feature absent from other "invisible" compressors) is invaluable for governing the uniformity of loudness.

Digital recording and compact discs mercilessly reveal the side-effects of conventional compressors; the Co-Operator's freedom from dynamic distortion readily meets the challenge of the digital age.

Audio and Video Production: Where time is money, the Co-Operator sets up quickly and easily to protect VTRs, ATRs, or cart machines from overload during transfer. Or, use the Co-Operator on mic channels—its smoothness and silence-gating guarantee uniform, punchy voice quality without noise rush-up during pauses.

Cassette Duplication: Even with the latest advancements, such as "hot" tape and Dolby® HX Pro, there are still some masters which can cause high-frequency saturation in cassette dupes. The Co-Operator's high-frequency limiter can be set to eliminate problems caused by synthesizers, cymbals, and sibilance, while still permitting high average modulation on the cassette. The broadband AGC/leveler can be defeated to permit use of the HF limiter alone when automatic gain riding or broadband peak protection is not desired.

Sound Systems: The Co-Operator provides colorless protection for your system—whether it be a fixed installation or traveling PA. For example, in an unattended bi-amped system, the slow AGC leveler can efficiently ride gain while the faster compressor protects the system from abrupt increases in level.

Placed after the mixer and equalizer and before an active two-way crossover, the Co-Operator will protect power amps from excessive clipping and high-frequency drivers from thermal overload.

Especially with constant directivity horns, the selectable pre-emphasis of the HF Limiter can protect the horn from the boost that is required to compensate for the constant-power high-end roll-off. The Co-Operator guards against over-heating and subsequent thermal failure.

In an attended bi-amped system, the Co-Operator can be placed *after* an electronic crossover. The two channels of the unit are arranged to separately feed the power amps for LF and HF drivers. In this application the leveler would be defeated (so it would not fight with the board operator), leaving the compressor, high-frequency limiter, and peak clipper to guard the system from overload. In particularly high-SPL environments, the compressor may be uncoupled so that each channel of the Co-Operator can operate independently to deliver the maximum level to each driver without distortion.

Similarly, the Co-Operator's controlled clipping avoids clipping of the power amplifier, which can cause "sticking", instability, and other audible problems.

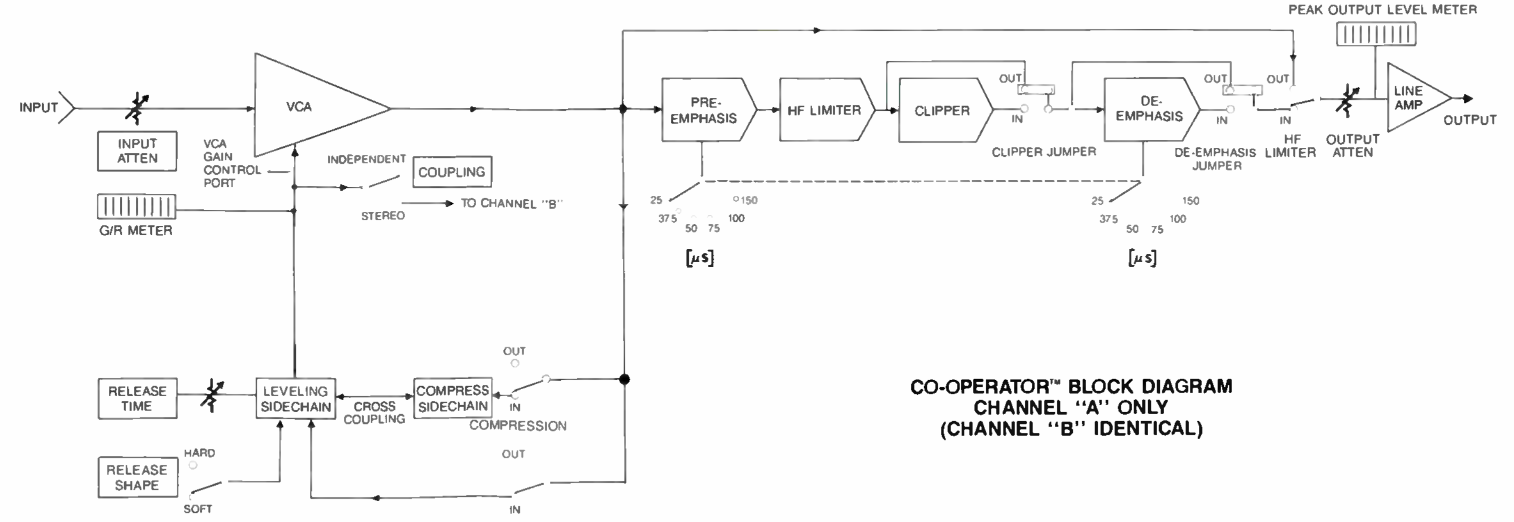
Broadcast Studio/Transmitter Link Protection: The STL is often the weak link in the broadcast chain due to barely adequate signal-to-noise ratio. Clean protective limiting ahead of the pre-emphasized STL transmitter helps maximize the signal-to-noise ratio at the receiver and prevents overload. Selectable pre-emphasis ensures matching of the high-frequency limiting function to the STL's overload characteristic.

FM Subcarriers: Voice or music SCA channels are typically pre-emphasized at 150µs. In addition to providing transparent gain riding and level control for the subcarrier, the Co-Operator can also provide the rarely found 150µs HF limiting function, which permits more efficient use of the SCA channel by eliminating overmodulation on HF peaks.

Circuit Features

The Co-Operator consists of a subtle AGC/compressor (AGC and compression can be used separately or together) cascaded with a HF limiter. To provide absolute peak control without the pumping associated with fast-attack limiter circuits, the HF limiter circuit includes a pre-emphasized clipper (which is jumper-defeatable). When the HF LIMITER switch is set to OUT, the entire HF limiter circuit (including its clipper) is removed from the signal path, ensuring maximum transparency.

Although AGC attack and release times adapt automatically to the nature of the program material to ensure the smoothest and least audible compression, the operator can fine-tune the Co-Operator to obtain the precise type of dynamic range control desired by adjusting the relative speed of the program-controlled release process. When the control of transient overshoots provided by the leveling function is insufficient, the faster "compression" function can be switched in to provide additional transient overshoot protection for material with abrupt level changes or unusually high peak-to-average ratios.



CO-OPERATOR™ BLOCK DIAGRAM
CHANNEL "A" ONLY
(CHANNEL "B" IDENTICAL)